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(54) Digitally multiplexed transmission system.

(57) The high performance multiplexed transmission system of this invention is configured by a sound coding unit (20) for coding voice input information by separating it into a core information part for assuring the minimum acceptable sound quality and a supplementary information part discardable in stages per the transmission priorities; a silence section detecting unit (21) for detecting silence sections of voice input information; and a multiplexing unit (22) for multiplexing only the information synchronizing with the correspondent's coder for the voice channels from which no sound is detected, or first core information part and second the supplementary information part from the ones with the highest priorities in stages in fixed length frames, for discarding the supplementary information parts which cannot be multiplexed because of a band deficiency.

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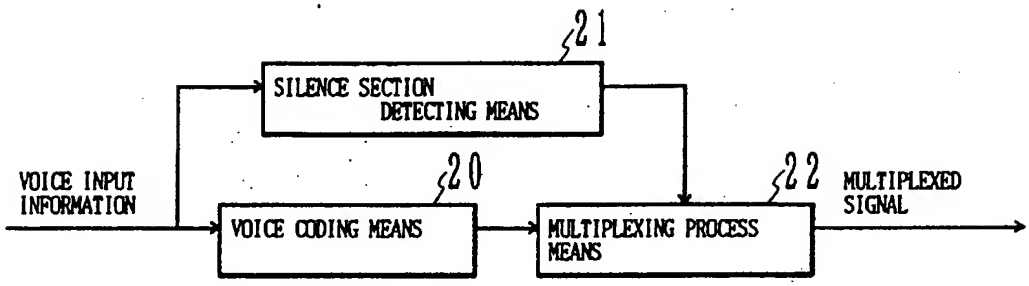

21
SILENCE SECTION DETECTING MEANS
VOICE INPUT INFORMATION
20
VOICE CODING MEANS
22
MULTIPLEXING PROCESS MEANS
MULTIPLEXED SIGNAL

Fig. 5A

Background of the Invention

This invention relates to a multiplexing device for sending multiplexed signals through a digital line, and to a high performance digitally multiplexed transmission system in a switching system.

Figure 1 shows an example of a voice multiplexing device or a switching system, and Figure 2 shows an example of a multi-media multiplexing device or a switching system. In Figure 1, voice data from a phone terminal 1 are applied to a relay transmission network 3 through a voice multiplexing device/switching system 2, and said voice data are transmitted to a receiving phone terminal 5 through a voice multiplexing device/switching system 4 on the receiving side. The relay transmission network 3 comprises a digital relay line and a voice multiplexing device/switching system for a relay.

In Figure 2, transmission information inputted to a multi-media multiplexing device/switching system 9 from the phone terminal 1 through PBX 6, from a data terminal through a packet switching system 7, and from a TV conference terminal through an image switching system 8 are applied to a relay transmission network 10. On the receiving side, data are outputted to a PBX 12, a packet switching system 13, or an image switching system 14 on the receiving side through a multi-media multiplexing device/switching system 11.

In such voice multiplexing devices, multi-media multiplexing devices, etc., a time-divisional multiplexing (TDM) method is usually adopted. However, another method such as a digital speech interpolation (DSI) multiplexing method or a digital data interpolation (DDI) multiplexing method may also be adopted as a statistical multiplexing method for efficient transmission. A DSI method detects a silence section in voice data and transmits only speech sections. A DDI method supplements information from queuing data communication media such as packet switches system before performing a multiplexing operation.

Figures 3A and 3B show examples of band allocations. Figure 3A is for a conventional TDM multiplexing method and Figure 3B is for a statistical multiplexing method. In Figure 3A, a band is allocated to each voice channel and the object is to get a higher performance voice coder with a view to improving sound quality within the allocated band. By contrast, in Figure 3B, bands are flexibly utilized by a plurality of voice channels, thus extending the equivalent band per channel and improving transmission quality.

Figure 4 shows a band allocation using a DSI method. In this method, a band is flexibly allocated depending on whether data comprise speech or silence by detecting silence sections. Where there are a greater number of voice calls more bands are allocated and sound quality is improved.

As described above, in a conventional TDM multiplexing method, for example, the same allocated to the maximum number of calls are also allocated to a small number of calls. The problem with this method is that sound quality is kept constant as for the maximum number of calls, and the average band cannot be extended by the flexible allocation of bands according to the voice level.

In the DSI method, silence sections are detected and no information of those sections is transmitted. That is, the band for those sections is considered to be zero, thus improving the transmission efficiency. However, when a line is congested, the whole coded information for one voice channel is either transmitted " as is " or discarded completely. This causes the problem that sound quality greatly deteriorates for a channel where the whole coded information is discarded completely.

Summary of the Invention

This invention pertains to a multiplexed transmission system for use in a voice multiplexing unit, a multi-media multiplexing unit, a PBX, a public voice switching network, and a multi-media public network, using a public or private digital communication line. It aims at statistically realizing both highly efficient multiplexing and high quality communication.

The system of this invention is configured by a sound coding unit, a silence section detecting unit and a multiplexing unit. The sound coding unit codes voice input information by separating it into a core information part for assuring the minimum acceptable sound quality and a supplementary information part discardable in stages according to transmission priorities. The silence section detecting unit detects silence sections of voice input information. The multiplexing unit multiplexes only the information synchronized with the correspondent's coder for the voice channels from which silence is detected, or first the core information part and second the supplementary information part from the ones with the highest priorities in stages in fixed length frames, for discarding the supplementary information parts that cannot be multiplexed because of a band deficiency.

Brief Description of the Drawings

Figure 1 shows an example of a voice multiplexing device or a switching system;
Figure 2 shows an example of a multi-media multiplexing device or a switching system;
Figures 3A and 3B show examples of band allocations using a conventional TDM multiplexing method and a statistical multiplexing method;
Figure 4 shows an example of band allocations using a DSI method;
Figures 5A through 5G are block diagrams of this invention;
Figures 6A and 6B are block diagrams for explaining the segmentation of voice transmission information;
Figures 7A, 7B and 7C are views for explaining the basic configuration and operation of an embodiment of the first principle of this invention;
Figures 8A, 8B and 8C are views for explaining the basic configuration and operation of the second embodiment of the transmission system in this invention;
Figures 9A, 9B and 9C show a third embodiment of this invention and views for explaining its operation of the transmission system;
Figure 10 is a block diagram for explaining the configuration of the fourth embodiment of the transmission system of this invention;
Figure 11 is a block diagram showing the basic configuration of the fifth embodiment of the transmission system of this invention;
Figures 12A and 12B show the sixth embodiment of the transmission system and a diagram for explaining the operation of this invention;
Figure 13 shows an embodiment of a multiplexing frame where a transmission band is transmitted as a code;
Figure 14 shows an embodiment of a method of discarding determination for the current transmission frame according to the past discarding history;
Figure 15 shows an embodiment of a voice multiplexing transmission system;
Figures 16A and 16B show a basic configuration of a multiplexing device and a basic example of the process of a multiplexing unit;
Figure 17 is a block diagram of an embodiment of a voice coder;
Figures 18A and 18B are views outlining the speech/silence determining method;
Figure 19 is a view for explaining an embodiment of the speech/silence mode determining process;
Figure 20 are views for explaining an embedded ADPCM coding method;
Figure 21 is a view for explaining the transmission route of voice and data in a multiplexing device;
Figure 22 shows an embodiment of a transmission band allocation code;
Figure 23 shows an embodiment of a code table of transmission band information;
Figure 24 shows an embodiment of serial data on the line multiplexed to a 64 kbps line;
Figures 25A and 25B show an embodiment of a frame configuration;
Figure 26 is a block diagram of an embodiment of a multiplexing unit;
Figure 27 is a flowchart of an embodiment of a general process of a multiplexing unit;
Figure 28 is a view for explaining the process cycle in a multiplexing unit;
Figure 29 is a flowchart indicating the first embodiment of the band arbitration process at the multiplexing operation;
Figure 30 shows an embodiment of arbitration based on the flowchart shown in Figure 29;
Figures 31A and 31B are flowcharts of a second embodiment of the band arbitration processes;
Figure 32 is a flowchart of an embodiment of discarding mode renewal processing;
Figures 33A, 33B and 33C show an embodiment of the arbitration according to the flowchart shown in Figure 31;
Figure 34 shows the relations among values of various functions in the band arbitration processors according to the flowchart shown in Figure 31;
Figure 35 is a block diagram showing an embodiment of a packet switching network to which this invention is applied;
Figure 36 is the block diagram showing the configuration of the packet interface part shown in Figure 35;
Figure 37 illustrates the operation of a speed difference absorption buffer;
Figure 38 illustrates the operation of an arrival deviation absorption buffer;
Figure 39 illustrates an embodiment of a packet formatting;
Figure 40 illustrates an embodiment of the code table of the transmission band information during sub-band coding;
Figure 41 is a flowchart of an embodiment of multiplexing using a sub-band coder;
Figure 42 shows an embodiment of the discarding mode corresponding to the frequency characteristics of voices;

Figure 43 shows an embodiment of the code table of the transmission band information for respective discarding modes;
Figures 44A and 44B are flowcharts showing an embodiment of the determination of the discarding mode and the mode notification to the receiving side;
Figure 45 is a block diagram of an embodiment of a compressed transmission system in which channels in a holding mode are assumed to be in a silence mode;
Figures 46A and 46B illustrate an embodiment in which a holding mode detector is contained in the switch;
Figure 47 is a block diagram showing an embodiment of a multiplexer containing a holding mode detector;
Figure 48 is a block diagram showing a schematic configuration of a system in which the multiplexer accommodates various transmission media;
Figure 49 is a block diagram showing an embodiment of a coder for coding according to the result of FAX protocol detection;
Figure 50 shows an outline of G3-FAX transmission procedures;
Figure 51 shows an embodiment of the additional steps to the flowchart shown in Figure 41 for enabling a discarding before a silence mode;
Figure 52 is a block diagram showing the configuration of a system capable of ignoring the result of detection or halting the operation of a silence mode detector;
Figure 53 is a block diagram showing a basic configuration of pipeline multiplexation in one multiplexing unit;
Figures 54A, 54B and 54C illustrate an embodiment of a pipeline multiplexing method;
Figure 55 is a flowchart of an embodiment of pipeline multiplexations in one multiplexing unit;
Figure 56 is a flowchart of batch demultiplexations;
Figure 57 is a flowchart of split demultiplexations;
Figure 58 illustrates the delay reduction according to the pipeline multiplexation method;
Figures 59A and 59B illustrate an embodiment of a voice quality protection method by simultaneously notifying a multiplexing device and a coder of voice detection in formation;
Figure 60A and 60B illustrate a method of generating noises on the receiving side according to the noise level on the sending side for silence periods;
Figure 61 is a block diagram showing the configuration of an embodiment of a call detection circuit;
Figure 62 shows an example of signaling information;
Figures 63A and 63B are flowcharts of an embodiment of a signaling transmission compression;
Figure 64 is a block diagram showing an embodiment of an image processing system to which this invention is applied; and
Figure 65 shows an example of determining the request band.

Description of the Preferred Embodiments

Considering the above described problems of the prior art technology, this invention is established to multiplex a core information part to ensure the minimum sound quality of a voice channel, to perform a multiplexing operation on supplementary information parts in the order of higher priority to obtain desirable sound quality, and to minimize the communication quality even during the congestion, thereby improving the communication quality in statistical multiplexing and multiplexing efficiency.

Figure 5 is a block diagram of this invention. Figures 5A to 5D show block diagrams for explaining the principle of a high performance digitally multiplexed transmission system in a multiplexing device or a switching system for transmitting a multiplexing signal through a digital line.

In Figure 5A, a voice coding means 20 is an ADPCM embedded voice coder for coding voice input information after dividing it into a core information part and a supplementary information part. In the core information part, the minimum sound quality is ensured when the information is transmitted, where the information is compressed and coded by differential information, etc.

The supplementary information part must be transmitted with the core information part to obtain desirable sound quality, and can be discarded sequentially, for example, every 10 bits, in the order of transmission priority of discard. An example of this information is supplementary information of the difference between a value predicted from core information and an input signal after being coded by a simple PCM quantizing unit.

A silence section detecting means 21 detects silence sections for using the result for transmitting information, as in a conventional statistical multiplexing method.

A multiplexing processing means 22 multiplexes, for fixed length digital slots repeated at a certain interval, only subordinate information (side information) necessary for synchronizing the modus operation to a correspondent's coder for a voice channel where the silence section detecting means 21 detects any silence sections among a plurality of voice channels to be multiplexed on a digital line, while it multiplexes said core information for a voice channel where the silence section detecting means 21 detects no silence sections.

Then, the multiplexing processing means 22 multiplexes the supplementary information part of the voice channel where no silence sections are detected for the remaining fixed digital slots. In this case, every 10 bits, for example, of the information is multiplexed in the order of higher transmission priority , that is, starting with the heaviest weighted part, and any supplementary information part that cannot be multiplexed due to a lack of bands is discarded.

Figure 5B shows a block diagram for explaining the principle of a multiplexing method based on a transmission request band for each voice channel. A request band determining means 23 determines the transmission request band of voice input information for each of a plurality of voice channels to be multiplexed on a digital line.

The multiplexing processing means 22, when discarding supplementary information of each voice channel where the silence section detecting means 21 detects no silence sections, performs a multiplexing operation for each voice channel by discarding every 10 bits, for example, of supplementary information parts within a request band determined by the request band determining means 23 in the order of lower transmission priority, that is, starting with order lightest weighted part.

Figure 5C is a block diagram for explaining the principle of a multiplexing method based on the call detection result. In Figure 5C, a call detecting means 24 detects whether or not a call is being made, that is, a calling mode or non-calling-mode, in each of a plurality of voice channels. The multiplexing processing means 22 performs a multiplexing operation on a voice channel in the calling mode without transmitting any coded information including side information, which is necessary for synchronizing to a correspondent's coder, for a voice channel where the call detecting means 24 detects the non-calling-mode.

When a multiplexing device or a switching system transmits signaling information for a voice channel with a signal-send (SS) signal and a signal-receive (SR) signal in a call detecting method, the calling mode or non-calling-mode in a voice channel is detected at 25a as shown in the operational block diagram in Figure 5E, according to the time period where the value of a logical OR or logical product of said two signals is kept constant while these SS and SR signals are monitored.

Figure 5D is a block diagram for explaining the principle of a multiplexing method when a multiplexing device or a switching system is connected to a label multiplexing network such as a packet network, an ATM network, etc. In Figure 5D, a multiplexing frame outputted by the multiplexing process means 22 is applied "as is", or after being divided into packets, to a label multiplexing network by a packet assembling means 25. Figure 5D shows a packet assembling means 25 in addition to the configurations shown in Figure 5A, but it can be added similarly to the configuration shown in Figures 5B and 5C.

In this invention, voice information each voice channel is segmented into a core information part and a supplementary information part, as described above, and then coded by the voice coding means 20. Figure 6 is a block diagram for explaining the voice transmission information segmented as described above. Figure 6A shows how a core information part and a supplementary information part are segmented. The core information part is compressed and coded using differential information, etc. and comprises core information for ensuring the minimum sound quality and side information necessary for synchronizing to the operating mode of a correspondent's coder.

The supplementary information part includes supplementary information to be transmitted after the difference between the value predicted from core information and an input signal is coded by a simple PCM quantizing unit. It is stacked sequentially on top of the core information part from the heaviest bit. Thus, it is sequentially discarded from the lightest bit, that is, the bit at the top. The core information part and the supplementary information part are transmitted where speech sections are detected, while only side information in the core information part is transmitted where silence sections are detected. Figure 5C shows that nothing is transmitted for the non-calling-mode.

Figure 6B shows a block diagram for explaining the band of requests to send data in each case shown in Figures 5A, 5B and 5C. In Figure 6B, if, in the core information part, 10 bits of side information and 20 bits of core information add up to 30 bits, and 10 bits each of core information make a total of 50 bits, 30 bits of the core information part (the sum of core information and side information) and 50 bits of the supplementary information part, adding up to 80 bits, are transmitted where speech sections are detected, while only 10 bits of side information is transmitted where silence sections are detected. During the discarding because of congestion, every lighter 10 bits of supplementary information are sequentially

discarded from data for speech section. As a result, the transmission band of each channel bands from 30 bits comprising only a core information part to 80 bits including all supplementary bits for speech sections, and indicates 10 bits comprising only side information for silence sections.

In a block diagram showing the principle of this invention as shown in Figure 5B, the request band of each channel for speech sections can be set in units of 10 bits to any value from 30 bits comprising only a core information part to 80 bits including all supplementary information. From the initial value of the request band, bits added for each channel are discarded sequentially in units of 10 bits at the discarding because of congestion, thus performing the arbitration of a band.

Figure 5C is a block diagram for explaining the principle of this invention where no 10 bits of side information are transmitted, that is, no coded information is transmitted, for voice channels where the call detecting means 24 detects the non-calling-mode. In the calling mode, only 10 bits comprising side information are transmitted for silence sections, and total data ranging from 30 bits of only a core information part for speech sections to 80 bits including all supplementary information are transmitted to each channel.

Next, in this invention, to further improve the transmission efficiency, various control data are compressed and transmitted. For example, when signaling information for a voice channel is transmitted as a signal-send (SS) signal and a signal-receive (SR) signal, (at 26a shown in the block diagram in Figure 5F), compressed transmission of signaling information can be performed where the SS signal remains unchanged, by transmitting a frame with a flag added to it indicating whether or not an SS signal is transmitted in a certain transmission cycle.

When a level signal is sent as a remote signaling (RS) signal for confirming the mode of a correspondent's device while comprising data channels, at 27a as shown in Figure 5G, the RS signal can also be compressed and transmitted at 27b with a flag indicating whether or not an RS signal is transmitted added to a frame.

As described above, in this invention, only side information in a core information part is transmitted for silence sections, while total data including the supplementary information part are transmitted for speech sections. During discarding because of congestion, lighter bits in the supplementary information part are sequentially discarded, thus improving the communication efficiency and permitting compressed transmission of various control data.

Figure 7 shows a view for explaining the basic configuration and operation of the first embodiment of the transmission system of this invention. In the block diagram of the basic configuration shown in Figure 7A, the embodiment comprises a voice coder 30 for coding input voice information provided by a phone, a silence section detector 31 for detecting silence sections in input voice information, and a multiplexing part (MUX) 32 for multiplexing the output of the voice coder 30 according to the detection result provided by the silence section detector 31 and then outputting the result to the line side. The information to be used by a multiplexing part 32 for multiplexing voice comprises coded data outputted by the voice coder 30 (a band for each channel is predetermined to 80 bits, for example) and the speech/silence mode of each voice channel outputted by the silence detector 31. The silence mode can be detected either by the voice coder 30 without independently providing a silence section detector 31 or by the multiplexing part 32 using the output from the voice coder 30.

A voice coded data length V_i' actually transmitted to each voice channel (expressed by subscript i) is given by the following expression:

$$W' \geq \Sigma \left(VDF_i \times \left(V_i' - V_s \right) \right) + N \times V_s \quad \cdots\cdots (1)$$

and

$$V_i \geq V_i' \geq Vmin \quad (2)$$

Here, Vi' is the number of bits assigned to each channel through a band arbitration process described later and corresponds to the band. The band of a channel is the product of the number of bits assigned to the channels multiplied by the number of frames transmitted per second, which is a transmission bit rate. To guarantee the minimum acceptable sound quality, the required number of lines must be determined at

the step of designing the line configuration by the following expression:

$$W' \geq N \times Vmin \quad (3)$$

where: W' is the number of bits available on the line side; V_l is the voice coded data length (predetermined as a fixed value); V_s is the number of bits of voice coded data (side information) to be transmitted even during silence sections; V_l' is the length of voice coded data to be transmitted; VDF_l is the speech flag; N is the number of channels to be accommodated; and Vmin is the coded data length necessary for transmission at minimum acceptable sound quality.

The voice coder 30 requires that data be coded in a deletable form within the range from V_l to V_{min}. For example, an embedded coder used in the adaptive difference (AD) PCM method is adopted.

Figure 7B shows the mode of a band and voice sound quality in the first embodiment. Detecting a silence section and transmitting only side information to the silence voice channel can reduce the deterioration of a band and of sound quality associated with the increasing number of voice calls. Even during the discarding because of congestion, there is only a very scant possibility of complete data for a specific channel being discarded temporarily, thus enabling the reduction of bands during the congestion to be minimized.

A DSI in "the improvement through a DSI" in Figure 7B indicating the sound quality shows statistical multiplexing, referring to the improvement of sound quality by extending the average band allocation.

Figure 7C shows an example of the multiplexing process during the discarding because of congestion in the first embodiment. In Figure 7C, channels #2 and #4 refer to the silence mode for which only side information is transmitted. During the discarding because of congestion, bits 5 in the supplementary information for respective channels in speech mode sequentially discarded starting from channel #0, then bits 4 in the supplementary information are sequentially discarded starting from channel #0 through to channel #3, and all the remaining data are multiplexed. The data are multiplexed starting with the side information of each channel as shown by 1 in Figure 7C, ending with the supplementary bits 4 of channel #5.

Figures 8A, 8B and 8C are block diagrams illustrating the basic configuration and operations of a second embodiment of the transmission system in this invention. In the block diagram Figure 8A showing the basic configuration, a request band determining part 33 is provided, in addition to the configuration components shown in Figure 7A, for determining the transmission band necessary for the voice information of each voice channel inputted from the telephone unit side.

The information used by the multiplexing part 32 for multiplexing voice includes coded data outputted by the voice coder 30 (The band can be either fixed or variable), the speech/silence mode information outputted by the silence section detector 31, and the voice coded data length outputted by the request band determining part 33 which is necessary for request quality transmission.

The speech/silence mode can be detected by the voice coder 30 and the multiplexing part 32. The voice coded data length required for requested quality transmission can be detected by the voice coder 30 and determined by the multiplexing part 32.

The multiplexing method in the second embodiment is determined in the identical manner to the expressions (1), (2) and (3) for the first embodiment.

However, the voice coded data length V_l is, unlike in the first embodiment, not predetermined as a fixed value, but a resultant data length obtained after the request band determining part 33 determines a request band for input voice information. The condition for the voice coder 30 is the same as in the first embodiment.

Figure 8B shows a voice band and sound quality based on the number of voice calls in the second embodiment. In the first embodiment described above, the voice transmission request level can be expressed in 2 values depending on the speech/silence modes. By contrast, in the second embodiment, the speech mode can further be divided into a plurality of levels to perform precise control, thus permitting further advanced improvement of sound quality.

Figure 8C shows an example of a multiplexing process in the second embodiment. In Figure 8C, channels #2 and #4 refer to the silence mode for which only side information is transmitted. For example, the request band for channel #0 is up to bits 3 in the supplementary information; the request band for channel #1 is up to bits 4 in the supplementary information; and the request band for channel #3 is up to bits 5 in the supplementary information. During the discarding because of congestion, every 10 bits of the upper supplementary bits in the request band in each channel are sequentially discarded. In Figure 8C, supplementary bits are discarded up to bits 2 in the supplementary information for channel #1, and all the remaining data are multiplexed.

Figure 9 illustrates a third embodiment of this invention and its operations. In Figure 9, a call detecting part 34 is provided in addition to the configuration components for the second embodiment shown in Figure 8. This is for detecting whether or not a call is being made, that is, the calling mode or the non-calling-mode, to prevent transmitting coded data to a voice channel having no calls at all. A switching system 35 is also provided between the phone unit side and the voice coder 30.

The information used by the multiplexing part 32 for voice multiplexing includes the speech/silence mode information detected by the call detecting part 34 in addition to the information in the second embodiment. However, the speech/silence mode can be notified by the switching system 35, or determined by either the voice coder 30 or the multiplexing part 32.

In the method for multiplexing voice, the following expression (4) is used instead of (1) in the first and second embodiments:

$$W' \geq \Sigma (VDF_i \times (V_i' - V_s)) + \Sigma (V_s \times CDF_i) \quad \cdots\cdots (4)$$

Expressions (2) and (3) are also used in the third embodiment. In expression (4), a CDF_i is a calling-mode flag for a channel i, set to 1 during the calling mode, and set to 0 in the non-calling-mode. The condition for the voice coder 30 is the same as that described above.

In this invention, the calling mode is not notified by a switching system, but detected by a coder or a multiplexing part, where a signal-send (SS) signal and a signal-receive (SR) signal are monitored as signaling information of voice channels to detect the calling/non-calling mode. Figures 9B and 9C show an embodiment of call detection. In Figure 9B, a call is detected by monitoring the SS signal sent from the sending side and the SR signal sent from the receiving side, where a mode notifying path from a switching system is not required.

Figure 9C shows a logic of call detection. In Figure 9C, assuming that SS and SR signal are set to 1 when a call is not made at all (idle), and set to 0 when a call is made (busy), if the logical sum of SS and SR signals remains at '0' for more than the specified time period τ_1, for example 1 second, it refers to the calling mode. Meanwhile, if the logical product of SS and SR signals remains at '1' for more than the specified time period τ_2, for example 1 second, it refers to the non-calling-mode.

Figure 10 shows a block diagram illustrating the configuration of a fourth embodiment of the transmission system of this invention. In this embodiment, a multiplexed frame generated by the multiplexing part 32 is assembled in a packet form of the label switching system network, for example in fixed length packet form and an ATM cell form by a packet assembling part 36, and then transmitted within the label switching system network.

Figure 10 shows the configuration of the packet assembling part 36 added to the first embodiment shown in Figure 7. It is obvious that a packet assembling part can also be added to the second and third embodiments. The operations of the voice coder 30, the silence section detector 31, the multiplexing part 32, etc. are the same as those in the first, second and third embodiments.

Figure 11 is a block diagram for explaining the basic configuration of a fifth embodiment of the transmission system of this invention. It shows a sample of the voice information of a plurality of voice channels which is inputted from the phone unit side through the switching system 35 and coded by the voice coder 30, and an embodiment of a multiplexing device for outputting said information to the line side after multiplexing by the multiplexing part 32, a packet and data inputted from a packet switching system or a data terminal, etc. through a switching system 37 and the accommodated interface 38. The switch system 35 and 36 are not required where a fixed route is set without requiring any switching system.

A significant information detecting part 39 detects the necessity for transmitting a packet or data inputted to the accommodation interface 38 by detecting an insignificant part having the value all 1 and the usage of a terminal. The detection can be performed by the switching system 37 or the multiplexing part 32. The information used for multiplexing media other than voice channels such as a data terminal and a packet switching system is transmission information generated by the accommodation interface 38 (a band can be fixed or variable) and the significant information for the transmission information.

As the multiplexing method, a band required for multiplexing a packet and data inputted through the switching system 37 or the accommodation interface 38 is subtracted from the number of bits W' available on the line side, and the band W'' allocated to a plurality of voice channels is determined by the following

expression:

$$W'' = W' - (\Sigma (DAF_i \times (D_i - D_{s-i}))) - \Sigma D_{s-i} \quad \cdots\cdots (5)$$

The number of available lines is determined by the following expression (6) so that the minimum sound quality can be ensured at the step of designing the line configuration,

$$W' \geq N \times Vmin + \Sigma D_i \quad (6)$$

For the voice information of each of the voice channels, one of the above described methods in the first to third embodiments is used for multiplexing. As described later, the history can determine the discarding method.

In expressions (5) and (6), W" is the number of bits available on a voice channel, N is the number of voice channels accommodated, D_i is the transmission data length of a packet or each datum, DAF_i is a significant/insignificant flag of data, and D_{s-i} is control information for each datum.

Figures 12A and 12B show a sixth embodiment of the transmission system of this invention and its operations. In the configuration diagram, Figure 12A, only an A/D transformer 391, a sub-band coding filter 392, an ADPCM embedded coder 393 to 396, and a multiplexing part (MUX) 397 are indicated. There is no silence section detector, etc. In Figure 12, input voice information is transformed to a digital signal by the A/D transformer 391, divided by the sub-band coding filter 392 into, for example, a band B1 of 0 - 1 KHz, B2 of 1 - 2 KHz, B3 of 2 - 3 KHz, and B4 of 3 - 4 KHz, each block of information being coded by respective ADPCM embedded encoders 393 through 396, multiplexed by a multiplexer (MUX) 397, and then outputted to the packet network side.

Figure 12B shows an embodiment of discarding level in a multiplexing device to which the sub-band coding method is applied. In Figure 12B where there are four sub-bands, it is assumed for simplification that core and supplementary information parts in respective sub-bands respectively have 2 and 3 bits. As shown in Figure 12B, at level 1, all bits in core and supplementary information parts are transmitted without being discarded. At level 5, 2 bits in the respective core information parts are transmitted to bands B1 B2 and B3, but the bits in the core information part for band B4 having the highest frequencies are discarded without being transmitted.

In this invention, as described e.g. in Figure 7C, the voice transmission band for each voice channel is split prior to transmission. However, a multiplexed frame on a line transmits the transmission band of each channel as a code. Various meanings depending on the kind of voice coder and the operating mode the setting are allocated to these codes.

Figure 13 shows an embodiment of a multiplexed frame where a transmission band is transmitted as a code. In Figure 13, an R_i, transmission band information for each voice channel coded following a synchronous flag F of the frame, is multiplexed first, and then transmission information for each actual channel is multiplexed. The transmission band code R_i is made error-resistant by adding an error correction code (ECC).

Described above is the whole procedure for efficiently transmitting, to a correspondent, the transmission band information necessary for holding voice channels in a variable band. With regard to media other than those for voice transmission, similar codes can be used for notifying of the transmission band, specifically of the necessity for transmission. For media such as data channels which provide fewer modes, the number of bits can be reduced, where the efficiency can be further improved by filing the data of a packet switching system in the blank.

In Figure 11 indicating the fifth embodiment described above where queuing media (e.g. packet switching system), other than voice channels, are multiplexed, a band can be reserved for the minimum throughput required for respective media. Then arbitration is made for voice multiplexing, and finally remaining bands are allocated to the media. Thus, the line line is efficiently utilized. In this case, the allocation of the minimum throughput to the above media can be made variable depending on a voice request band and activity (speech/silence mode, calling mode).

Next, an explanation is given with regard to the first through sixth embodiments, about the method of discarding information in consideration of the weight information for discarding according to the history and

the priority of each channel when coded voice information has to be somehow discarded. In arbitrating the discarding priority among channels, the number of accommodated lines is determined by the following expression to ensure the minimum request sound quality for each channel:

$$W' \geq \Sigma (Vmin\text{-}i) \quad (7)$$

For the actual multiplexing, the voice coded data length V_i' can be determined to meet following expressions (8) and (9):

$$W' \geqq \Sigma (VDF_i \times (V_i' - V_s)) + \Sigma V_s \times CDF_i \quad \cdots\cdots (8)$$

$$V_i \geq Vi' \geq Vmin\text{-}i \quad (9)$$

The meanings of the signs used in expressions (7) (8) and (9) are the same as those in the second and third embodiments. However, the coded data length Vmin required for transmitting information in the minimum sound quality is the same, while in this method, the value Vmin is set separately for each channel.

Figure 14 shows an embodiment of a method for determining the discard in the current transmission frame according to the past discarding history. In Figure 14, as shown in expression (1), restoration to the request band is performed sequentially when discarding because of congestion occurs. As shown by expression (2), a narrower band is allocated when the mode changes from silence to speech. In Figure 14, the arrow shows the time at which band allocation is determined.

Figure 15 shows an embodiment of a voice multiplexing transmission system. Figure 15 is almost identical to a part of the multi-media multiplexing transmission system as shown in Figure 2. Identical parts are allocated the same numbers. In the telephone units and extension system from the telephone unit 1 to the multiplexing device 9, analog or digital signals are used. In the relay trunk system from the multiplexing device 9 to the multiplexing device 11 through the transmission network 10, digitally coded transmission is performed.

Figures 16A and 16B show a basic configuration of a multiplexing device and a basic example of the process of a multiplexing unit. Figure 16A also refers to a configuration of a multiplexing transmission system comprising a multiplexing device, and multiplexing devices 40 and 41 are connected to the both ends of a transmission line 42. Each multiplexing device comprises a CODEC 43 corresponding to each voice channel and a multiplexing unit (MUX/DMUX) 44.

Figure 16B shows a basic example of the process of a multiplexing unit. In this invention, the embedded multiplexing method is adopted where the voice coded information is split for transmission into core bits and supplementary bits. Core bits refer to basic information that cannot be discarded, while supplementary bits refer to redundant bits that can be discarded when the congestion of voice information occurs in a multiplexing unit. Discarding supplementary bits necessarily deteriorates sound quality, but permits transmission of core bits, thus ensuring the minimum sound quality provided by core bits.

Since supplementary bits are sequentially discarded during the discarding because of congestion, the deterioration of sound quality can be minimized at the lower level of congestion. Besides, when a communication is not made or silence is detected in information being transmitted, a silence compression is made, and silence sections are not transmitted to the transmission line.

In Figure 16B, channel 3 is silence compressed for silence, and its core bits are not transmitted. The suppression levels of supplementary bits are arbitrated according to the congestion level. The supplementary bits of channel 2 are not transmitted at all, but the supplementary bits of channels 1 and 4 are transmitted.

Described below in association with Figures 16A through 16D are the functions required to execute the statistical multiplexing method of this invention, that is, the statistical multiplexing method applied to the activity fluctuation among voice channels transmitted to the same transmission line. In the frame format as shown in Figure 16B, the existence of transmission information of each channel, the emission band of transmission information of each channel, etc. are transmitted as a part of header information.

First, the method for identifying the activity of each voice channel and the notification route are

described. A coder of a CODEC 43 in a multiplexing device 40 shown in Figure 16A identifies the calling mode, the speech/silence mode, and the measuring information for transmission request, and the result is sent to a multiplexer (MUX) of a multiplexing unit 44. Here, the calling mode is identified by monitoring the signaling information, the speech/silence mode is identified by a voice detector (VDET) described later, and the weighting information for transmission request generates information of up to 8 levels/channels using the SN ratio as a yardstick.

A multiplexer in a multiplexing unit 44 arbitrates the line multiplexing based on the above described information. A band may be reduced to less than a requested band as a result of the arbitration. All coder information is transmitted to a multiplexer, and sometimes transmission beyond the requested band is enabled. After the arbitration of the line multiplexing, a multiplexer informs a demultiplexer (DMUX) of the multiplexing unit 44 within the correspondent's multiplexing device 41 through the transmission line 42, for example, the transmission level (code) of the identification result of calling, the speech/silence mode, and the arbitration result by a multiplexer.

The demultiplexer informs a decoder of a CODEC 43 within a multiplexing device 41 or a relaying multiplexer in the succeeding stage of received information "as is", and the decoder in turn recognizes the amount of coded data to be provided through the information from the demultiplexer of the transmission level, and then detects an occurrence of a discard.

Other requisite functions in the statistical multiplexing method of this invention include a relay switching function of the system, countermeasures to erroneous transmission on a transmission path, the utilization efficiency of a transmission path, and a function for monitoring the fluctuation of transmission quality of each channel.

As described above, the first function necessary for adopting an embedded multiplexing method is a redundancy detecting function of voice transmission, where the calling/non-calling-mode and the speech/silence mode are identified by a coder, and necessary transmission bands are informed to a multiplexing unit.

The second function is an arbitrating function for an embedded multiplexing, where the multiplexer in a multiplexing unit deletes unnecessary coded information and the remaining bands are flexibly allocated to other voice channels.

The third function is a discard information notifying function where a multiplexer notifies multiplexing units of the information on transmission bands of respective channels, and the information is provided as header information of periodically transmitted fixed length frames. An error correction code is added to a header as a countermeasure to a line error.

The fourth function is a logging function realized by a multiplexer and an external console, where the utilization of lines and the fluctuation of sound quality of each channel can be confirmed from an external console.

Figure 17 shows a block diagram of an embodiment of a voice coder. In Figure 17, a voice coder uses the following devices for sending data: a 6-line interface 45 comprising 4 lines for transmitting voice signals and 2 lines (SS/SR) for transmitting signaling information such as on dial pulses, etc., a call detector (CDET) 46 for detecting a call through monitoring by SS and SR signals, a multiplexing part (MUX) 47 for sending an internal bus interface, an internal bus interface circuit 48 for emission, an analog-to-digital (A/D) converting part 49 for transforming voice codes to digital signals, a coder (CODER) 50 for outputting voice codes identifying core bits that cannot be discarded and supplementary bits that can be discarded sequentially, and a voice detector (VDET) 51 for detecting silence sections by using the output from the A/D converting part 49.

On the other hand, the following devices are used for receiving data: an internal bus interface circuit 52 for receiving data, a demultiplexing part (DMUX) 53 for receiving an internal bus interface, a voice signal decoder (DECODER) 54 for decoding coded data, a noise mixing part (NMIX) 55 that is a noise interposing circuit for reducing abnormal impression given by silence sections, a digital-to-analog(D/A) converting part 56, and an OD circuit interface 57.

Figures 18A and 18B are views for outlining the speech/silence mode determining method. Figure 18A shows a part associated with speech/silence mode determination by the voice detector (VDET) 51 for detecting a silence section in the voice coder shown in Figure 17. An A/D converter 570 corresponds to the combination of A/D converting part 49 and D/A converting part 56 shown in Figure 17.

Figure 18B shows a view for describing the method of determining the speech/silence mode. The voice detector (VDET) 51 determines the speech/silence mode by cutting the strings of voice data coded by the A/D transformer 570 in a specific time unit, and then analyzing the voice data strings.

Figure 19 shows a view for explaining an embodiment for the speech/silence mode determining process. As shown in Figure 19, a voice signal is applied to a high path filter (HPF) 580 to clear the direct

current offset from a transmission voice signal, and the following two processes are performed on a high frequency signal.

In the first process, 581 calculates the electrical power for an input signal, 582 determines whether or not the electrical power exceeds a threshold, and a mode determining part 585 detects the speech mode when the electrical power exceeds the threshold.

In the second process, 583 counts zero-passings per sub-frame split from one whole frame, that is, the count of signals passing the zero level, 584 compares the count with a threshold, and the mode determining part 585 detects the speech mode when the zero-passing count exceeds the threshold.

That is, when the speech mode is detected in the first or second process, the result of the determination shows "speech", but the first process means determines "speech" as a large voice, and the second process determines a voice starting with a sound with a high frequency component as a word head.

In Figures 17, 18A, 18B and 19, the silence mode is detected by the voice detector (VDET) 51 connected to the voice coder (CODER) 50. However, the multiplexing unit 44 shown in Figure 16 can determine the speech/silence mode. In this case, the multiplexing unit 44 has to recognize the coded information and identify the silence mode. The simplest method of realizing this function is to provide a multiplexing unit with a decoding part of a voice coder and the voice detector, where coded information is first decoded and then searched for the silence mode like in a voice coder.

The following description explains how to realize the request band determining part 33 in the second embodiment of the transmission system described in Figure 8. The voice coder 30 includes the functions of the request band determining part 33 as described above. A voice coder in this invention is operated by the prediction coding where a coding part and decoding part synchronize to each other to operate an prediction unit. The coding part, quantizes the difference (prediction error) between an input signal and the prediction by the prediction unit.

Coding part to which the above principle is applied easily generates a signal equivalent to the signal re-generated by a decoding part, which is commonly called a local decoder. A local decoder calculates the electrical power error between an input signal and a re-generated signal provided by a local decoder for every specific samples; the result is divided by the electrical power of the input signal, and the resulting "segmental SNR" is calculated for each transmission band levels. The minimum band level that meets the specific SNR value is determined as the transmission request band. The relation between the segmental SNR and the transmission band level is described later.

On the other hand, when a request band determining part is provided separately from a voice coder, the involved process is identical to that in the silence detector. That is, the voice detector (VDET) 51 simply determines 1/0.

The determination is divided by levels, and each corresponding band is determined as the request band. For example, four-level determination can comprise complete speech, incomplete speech, incomplete silence and complete silence levels. When a request band is determined by a multiplexing unit, as in the above described silence detection, a voice decoding process is performed in a multiplexing unit and then a process similar to that in the determination by the voice coder is performed.

Figures 20A and 20B show views for explaining embedded ADPCM coding methods. Figure 20A shows a block diagram of a configuration of a common ADPCM coding method. Figure 20A shows an example of a conventional high compression coding method. For the input voice information, a subtracter 590 obtains the difference between the input value and the output of an prediction unit 593, and this difference is coded by a quantizing unit 591. Thus coded information is outputted. The coded information is applied to an prediction unit 593 by a dequantizing unit 592 for an adaptive prediction. This method is based on the synchronization of the sending and receiving sides. When data are discarded, the silence process is required.

Figure 20B is a block diagram of an embedded ADPCM coding method. This method differs from the common ADPCM method shown in Figure 20A on the following points: supplementary bits are deleted from the coded information (the sum of the core information and the supplementary information outputted from the quantizing unit 591) by a bit deleting part 594, and only the resultant core bits are applied to an prediction unit 596 through a dequantizing unit 595. That is, the internal loop forming an prediction signal passing through an prediction unit 596 contains only core bits, and the bits actually involved in the ADPCM coding are core bits only. "The compressed coding in a core information part using difference information" means this internal loop, and the prediction value of core information means the output from the prediction unit 596.

In the embedded ADPCM coding method, the quantizing unit 591 on the sending side has a resolution corresponding to the sum of the number of core bits and the number of supplementary bits. The dequantizing unit 595 has resolutions corresponding to the number of core bits only.

On the other hand, on the receiving side, a dequantizing unit having resolutions corresponding to the number of supplementary bits received is provided for the decoding. This means that supplementary bits are, in equivalence, simply PCM coded. That is, core bits are involved in the difference coding where a coded part and decoded part travel through the same prediction loop, which means supplementary bits are simply PCM coded because they are not used in the prediction loop.

Figure 21 is a view for explaining the transmission route of voice and data in a multiplexing device. The following description is given in association with Figure 21 for illustrating activity identification of voice channels and the notification path. In ① and ②, voice information in analog signals and signaling information in dial pulses of 10 PPS, for example, are given from a PBX 60 to a voice LS (line set, or terminal container) 61. Based on the information described above, voice signals are highly compressed and coded. For example, the signaling information is binary coded by sampling at 0.4 kbps, and then the voice activities are detected.

The voice activities are divided into the identification of the calling mode by monitoring the signaling information, the identification of the speech/silence mode by a voice detector, and the weight information for transmission request generated by a coder using the SN ratio as the yardstick. The method of identifying the calling mode by monitoring signaling information is illustrated in Figure 9.

Next, in ② through ⑦ , the activity information detected by the voice LS 61 to a MUX 64 is notified through relay switching system units (EXU) 62 and 63 in ③ through ⑥ using the code format (R_i code) described below.

The MUX 64 arbitrates the line multiplexing based on the activity information (R_i code) provided for each channel, and the result is given to the DMUX in the 3-bit-code format. As described later, the transmission band allocation code is given in 4 bits. However, the remaining 1 bit is used for compressed transmission of signaling information. In this case, the identification of the calling mode and the speech/silence mode can be determined by the information provided by the voice LS 61 , with the transmission level determined according to the arbitration result of the MUX, and can be sent to the MUX' 64 on the receiving side in ⑬ via lines 65 and 66 in ⑨ through ⑫.

The DMUX in the MUX' on the receiving side sends the transmission information based on the arbitration result in ⑭ through ⑲ to the voice LS 61 through the EXU 62 and 63, ⑮ through ⑱, or to the relay MUX for the next stage. The voice LS 61 recognizes the size of the voice coded information provided by the voice LS on the sending side according to the activity information obtained from the DMUX. Based on this recognition result, the voice decoding process can be performed. At this time, the occurrence of discard in the relay stage can be detected, and the decoded result is sent to the PBX 60 in ⑲ and ⑳. A CODEC 43 shown in Figure 16 is placed within the voice LS 61 in Figure 21, and the voice coder shown in Figure 17 corresponds to the content of the voice LS 61.

A plurality of a node control parts 67 are provided in a multiplexing device for controlling a plurality of the MUXs 64. These this node control parts store, multiplexing parameters which are determined according to the features of the coders for respective voice channels and the deterioration of sound quality due to the discard of supplementary bits. This parameter is set to the MUX 64 at system startup. These parameters can be used based on the features of coders where, for example, the sound quality can be reduced even if only core bits are transmitted when the speech mode is restored from the silence mode. Furthermore, a data LS 68 for a data channel and an interface 69 for a packet switching system 70 are provided in a multiplexing device.

Described below is the process for realizing a significant information detecting part 39 in the fifth embodiment of the transmission system shown in Figure 11. The "insignificant" determination is made simply when the data in the multiplexing process frame are all "1"; a flag indicating that the data within the frame are not transmitted (an idle flag described later) to the receiving side; on the receiving side, an idle flag confirms the existence of transmitted data; and all the "1" information is generated for a frame having no emitted data, thus suppressing the emission of unnecessary information.

In Figure 21, a significant information detecting part is embedded in the data LS 68 or the multiplexing unit 64.

When the detection is performed by the data LS 68, data in one cycle frame of a multiplexing process are stored in a memory or a shift register, and a comparator determines whether or not the stored values are all "1". Otherwise, a processor is equipped in the data LS 68, where it is determined whether the data in one cycle frame of a multiplexing process are all "1".

Next, when the detection is performed by a multiplexing unit, as in the data LS 68, either a memory or a shift register and a comparator are operated for the detection in a few channels, or a processor 77 in Figure 26 described later makes a determination through software processing. The latter requires a smaller scale hardware configuration.

Figure 22 shows an embodiment of a transmission band allocation code. In Figure 22, the transmission band allocation code is given in 4 bits, where the lower 3 bits are used for transmitting coded transmission band information (RI_2, RI_1 and RI_0) and the most significant bit b_3 is used for emitting signaling transmission flag. This signaling transmission flag is set to 1 for the transmission of signaling information, and to 0 for the compression of information. The signaling information is transmitted to the frame where signaling information is changed, but when signaling information remains unchanged, the signaling information received last by the DMUX on the receiving side is outputted to a voice channel.

For a data channel, a remote signaling (RS) signal transmission flag is transmitted as "1" for transmission and as "0" for compression to the most significant bit. The meaning is the same as that of a signaling transmission flag. An idle flag is sent to the least significant bit to indicate whether or not it is necessary to transmit data areas. The flag is set to "1" for transmission and "0" for compression. The intermediate bits b1 and b2 are filled with data from the packet switching system.

Figure 23 shows an embodiment of a code table of transmission band information. In Figure 23, the transmission band information RI_2, RI_1 and RI_0 are transmitted after being transformed to 3-bit codes depending on whether the state is non-calling, silence, or speech modes (6 steps). This transmission band information is multiplexed and transmitted after the arbitration. The transmission level is divided every 2 kbps to levels 1 through 7.

Next, the determination algorithm for determining the transmission request level by a local decoder using a segmental SNR in a voice coder is described in association with the levels shown in Figure 23.

In this determination algorithm, segmental SNRs corresponding to respective transmission levels 2 through 7, that is, an SNR (2) through an SNR (7), are calculated in every multiplexing frame cycles first. Next, when the determination threshold SNR (TH) of an SNR from the SNR (2) in the ascending number of levels is detected as larger than the threshold through the comparison with 25dB, the level is considered to be the transmission band request level. After comparing up to the seventh level, the level 7 is considered to be the transmission band request level when all segmental SNRs are smaller than the threshold SNR (TH).

Figure 24 shows an embodiment of serial data in the line multiplexed to 64 kbps line. In Figure 24, one frame comprises 640 bits, and the transmission time is 10 ms. The 640 bits are divided into phases 0 through 3, that is, 4 phases of 160 bits each. The transmission time for one bit is 15.6 μs.

Figures 25A and 25B show an embodiment of a frame configuration. Figure 25A shows a general configuration of a frame comprising 4 phases (0 through 3) where the leading 11 bits of respective phases (a total of 44 bits) are used as header parts and the bits including and following the 12th bit (a total of 596 bits ending with the 160th bit) are used as data multiplexing parts.

Figure 25B shows the configuration of the header parts. In Figure 25B, the header parts comprise 11 bits for respective phases, but the first bits F1, F2 and F3 for phases 0, 1 and 2 are used as frame synchronous patterns, and SEND for phase 3 is used as the device alarm bit. The second through the ninth bits (a total of 32 bits) in respective phase are used as 4 bits each of transmission band allocation code R_i - (#k) or for transmitting a packet PKT from a packet switching system. The tenth and eleventh bits in respective phases are used as storage areas for error correction codes ECC for transmission band allocation codes R_i (#k).

Figure 26 shows a block diagram of an embodiment of a multiplexing unit. In Figure 26, the multiplexing unit comprises a bus interface circuit 71 as an inter face to each channel, a bus interface circuit 72 as a line interface, a two-port RAM 73 provided with transmission request information by each channel and outputting multiplexed frame information, a write control part 74 for the two-port RAM 73, a RAM 75 for storing programs and data, an input interface circuit 76 for a packet channel, an operation process processor 77 for multiplexed and demultiplexing, an output interface circuit 78 for a packet channel, a two-port RAM 79 for storing received data, a write control part 80 for a two-port RAM 79, a multiplexing frame aligner 82 as a buffer circuit for synchronizing to a multiplexing cycle the transmission request information from each channel, an MSYNC part 83 as a cycle timing detecting circuit, a frame aligner 81 as a frame synchronization establishing circuit among MUXs, and a SYNC part 84 as a frame timing detecting circuit.

In Figure 21 , a node control part 67 sets a multiplexing parameter to each multiplexing unit 64 as described above, and the setting is performed through a control bus for a package of each multiplexing unit 64.

In Figure 26, such a control bus is skipped, but actually, for example, a register where setting information is written from the node control part 67 is connected to a memory bus of the processor 77; and the processor 77 in the process of executing the multiplexing program reads the content of the register if necessary.

Figure 27 is a flowchart outlining the processes in an embodiment of a multiplexing unit. In Figure 27, after the queue-time-out timer is activated at S85, the multiplexing process (MUX) is performed. In the

multiplexing process, transmission request information of each channel is received at S86, that is, the information is read from the two-port RAM 73 shown in Figure 26; the band arbitration for voice channels is performed at S87; a transmission band allocation code (R_i code) is generated at S88; an error correction sign is generated at S89; a transmission path frame is generated at S90; and a transmission path frame is emitted at S91, that is, information is written to the two-port RAM 73.

During the demultiplexing process (DMUX) of received data, a transmission path frame is received at S92, that is, read from the two-port RAM 79 shown in Figure 26; the correcting process is performed at S93 using error correction signs; a transmission path frame is demultiplexed to data of each channel at S94; information received at each channel and a notification frame of transmission band allocation codes (R_i codes) are sent, that is, written to the two-port RAM 79; and a timer starts for the next process cycle.

The band arbitration process at S87 is performed by a processor 77 shown in Figure 26. The arbitration program is either stored in a RAM 75 or embedded in the processor 77 as a mask ROM.

Figure 28 is a view for explaining the the process cycle. The process cycle T is a constant value, for example 5 ms or 10 ms, according to which a timer indicates a time-out every specific process cycle. During the cycle, the multiplexing (MUX) process and demultiplexing (DMUX) process are performed. The remaining time is considered idle, i.e. the time-out waiting time of the timer.

Figure 29 is a flowchart indicating a first embodiment of the band arbitration at a multiplexing operation. Figure 29 shows the band arbitrations corresponding to the first embodiment of the transmission method shown in Figure 7, the second embodiment shown in Figure 8, and the third embodiment shown in Figure 9. In Figure 29, as shown in Figure 6, the side information during the process cycle T is given in 10 bits, and the core information in 20 bits, thus forming the core information part having a total of 30 bits, and the supplementary information parts consist of 10 bits each, thus forming a total of 50 bits.

The transmission speed is 64 kbps; the multiplexing frame generating cycle is 5 ms; and the number of accommodated channels is 8. For a frame format, the number of bits allocated to voice channels BW is 260 excluding 4 flag bits, transmission band allocation code 4 x 8 bits = 32 bits, error correction signs ECC 8 bits, and signaling information 2 x 8 bits = 16 bits. The coded data length Vmin which is necessary during voice transmission in the minimum sound quality is 30 bits.

That is, the initial value (request band) of the voice coded data V_i (corresponding to V_i' in expression (1)) of each channel is 10 bits during silence in the first embodiment shown in Figure 7, and 80 bits during speech, totaling 2 levels only. However, it is 10 bits during the silence mode in the second embodiment shown in Figure 8, and variably between 30 and 80 bits in the speech mode; 0 bits in the non-calling-mode in the third embodiment shown in Figure 9, 10 bits in silence and variably between 30 and 80 bits in speech during the calling mode.

When the process begins as shown in Figure 29, the sum of the initial values of the voice code data length for the respective channels, e.g. the sum of the bands V_i requested by the respective voice channels, is allocated to the 8 channels in S100. LBW is obtained by subtracting the sum from the BW bits allocable to the voice channels, which are 260 bits in this case in S101. It is judged in S102 whether the LBW value is positive, 0 or negative. When the LBW value is positive or 0, since the request bands are totally satisfied, the process ends without an arbitration.

Since the band is deficient when the LBW turns out to be negative in S102, it is judged in S103 whether the voice code data length for channel j next to the channel containing bits last discarded by an arbitration during the last frame transmission is smaller than, equal to, or greater than the code data length V_{min} for transmission at the minimum acceptable sound quality. Since no more data for the channel can be discarded when V_j is smaller than V_{min}, the value of j is incremented in step S105, and the processes from S100 are repeated.

When V_j is larger than V_{min} in S103, 10 bits of the voice data for the channel are discarded in S104. After the value of j is incremented in S105, the processes after S100 are repeated. When LBW becomes non-negative in S102, all processes are terminated.

Figure 30 shows an embodiment of arbitration based on the flowchart shown in Figure 29. In Figure 30, T = 1 through 15 indicate arbitration timing when multiplexed frames are created. At T = 1, the bands requested by all channels 1 through 8 correspond to 10 bits, i.e. a silence mode. Respective channels receive the 10 bits as requested.

At T = 2, the bands requested by respective channels are all 50 bits. Since their sums, 400 bits, are greater than the earlier described BW 260 bits, some data need to be discarded. For instance, if the discarding begins from channel 1, 10 successive bits in channels 1 through 8 are discarded. Then, 10 successive bits in channels 1 through 6 are also discarded. Thus, a total of 140 bits are discarded and the arbitration is terminated.

Similar arbitrations are performed at and after T = 3. The voice data discarding at T = 3 starts from 10

bits in channel 7, i.e. the next channel after the one from which the last discarding was made. As a result, channels 7, 8 and 1 through 4 discard a total of 40 bits and channels 5 and 6 discard a total of 30 bits, before the arbitration is completed.

Although similar arbitrations are performed at following points in time, there are exceptions. Channel 8 discards only 30 bits at T = 7 and 8 because those 30 bits represent the core bits. Channel 8 does not discard any bits at T = 9 because side information during a silence should not be discarded.

Figure 31 is a flowchart of a second embodiment of this invention for the band arbitration processes. Figure 31 shows an embodiment of a processing for determining the method of discarding current transmission frames from the past discarding history. In Figure 31, the number of bits allocable to voice channels, the number of accommodating channels, the tag information part, and the core information part are the same as those in the first embodiment shown in Figure 29.

When the processes begin as shown in Figure 31, i (indicating the channel number) is set to 1 in S106, and subroutine DISCMD (Discard Mode Renewal Subroutine) is executed in S107. This is described later as the renewal of the discarding mode. Then, the value i is incremented in S116, and it is judged whether or not the value i is smaller than 8 in S117. If it is smaller, the processes from S107 are repeated.

Figure 32 is a flowchart of S107 shown in Figure 31, i.e. an embodiment of the discarding mode renewal processing. Figure 33 shows an embodiment of the arbitration, based on the flowchart shown in Figure 31. Figure 34 shows an embodiment of the relations among the values of respective variables. The flowcharts shown in Figures 31A, 31B and 32 are explained by referring to the embodiment shown in Figure 34.

As in Figure 30, which shows the first embodiment of arbitration, Figure 33A shows the bands requested by respective channels and those allocated as a result of arbitration processing, at respective arbitration timings T = 1 through 15. The first row of the display for respective channels at respective arbitration timings has the requested bands on the left side and the allocated bands on the right side. In the second row, the leftmost side indicates a discard mode flag DF, the center indicates a discard damage factor DMG, and the rightmost side indicates a margin gap factor GAP for the minimum allocated bands.

On the right side of these arbitration results for channels 1 through 8, all the requested bands and allocated bands are indicated. Among the requested bands, the right side indicates the total of the requests from the encoders for the respective channels, the left side indicates the total of the requested bands, taking a later described discard mode into consideration.

For instance, at T = 1, although the total of the requests from the encoders for the respective channels, 80 bits match the total of the requested bands considering the discard mode, since the speech mode is restored from the silence mode at T = 2, lower bands are allocated.

Here, the bands requested by respective channels are assumed to comprise the 30 bits of core information, and they total 240 bits immediately after the sound restoration, so that only 30 bits of the core information part are allocated to respective channels at T = 2.

Figure 33A shows the values of the discard mode flag DF, a discard damage factor DMG, and a margin gap factor GAP for the minimum allocated bands after the arbitration processings through the processes shown in Figures 31A, 31B and 32. Here, GAP = 9 indicates that GAP has a negative value. The flowchart shown in Figure 33B explains how these values are determined.

Figure 33B shows how many bits are actually allocated to the bands in response to the requests from channel encoders at arbitration timings T = 1 through 11. The channel herein is equivalent to a channel in Figure 33A.

At the arbitration timing T = 1 shown in Figure 33B, the discard mode renewal is performed in S107 shown in Figure 31. That is, in ST108 shown in Figure 32, the requested bits $V_I = 80$ bits are used to obtain the GAP. The coded data length V_{imin} necessary for transmitting voices at the lowest acceptable voice quality is set to 30 bits. As a result, GAP becomes 5, and the next step S109 checks whether or not the requested bits V_i are smaller than V_{imin}.

Since in this case V_i is larger, it is judged whether or not the discard mode flag is 0 in S110. Here, the discard mode flag DF indicates 1 for discarding and 0 for preserving. Since this is a case of the first arbitration timing, if the discard mode is not yet entered, the flag value remains 0 and the discard mode renewal processing is terminated and S116 shown in Figure 31 is invoked. If it is judged that the discard mode renewal for all 8 channels have already been performed in S117, the processes after S118 begin.

The processings in S118 through S124 in Figure 31 are essentially similar to the the processes shown in Figure 29 showing the first embodiment of the arbitration processing. That is, if the sum of the requests from the respective channels is larger than the bit number BW allocable to the voice channels, the arbitration processes cause the bands in the respective channels to discard 10 bits respectively and the process in S125 to be invoked. However, unlike the case shown in Figure 29, the channels which discard 10

bits in S123 increments their discard damage factor DMG values.

As shown in Figure 31, after the channel number i is set to 1 in S125, it is judged in S126 whether or not the DMG value is positive. Since the arbitration has only begun at T=1, as shown in Figure 33B, if the DMG values are 0 then, it is judged in S128 without having to go through S127 whether or not the DMG values are 0. Then, the discard mode flag DF is set to 0 in S129, the value of the channel number i is incremented in S131. It is judged in S132 whether i is smaller than 8. If i is smaller than 8, the processes from S126 are repeated, until i becomes equal to 8. The processes are then terminated. Here, the processes from S118 to S124 assume that the requested bits for the channels are not decreased. Consequently, 80 bits of V_i' are allocated to the channel.

A silence section starts at an arbitration timing T=2, as shown in Figure 33B, and 10 bits are requested. As a result, the GAP value in S108 in Figure 32 becomes -2, and it is judged in S109 that the requested bits are smaller than V_{imin}, and it is judged in S114 whether or not the change history from the silence mode to the speech mode should be referred to. Here, as illustrated in (2) of Figure 14 in order to lower the allocated band when the silence mode changes to the speech mode, the DMG value is changed in S115. Here, when the V_{imax} value is set to 80 bits, the DMG value becomes 6, and S116 shown in Figure 31 is invoked.

In the processes from S118 to S124 shown in Figure 31, the bits allocated to the channels are not decreased. It is judged in S126 whether DMG is positive or negative. Since the DMG is 6 at this time, the value is decremented in S127, it is not judged to be DMG=0 in S128, and the discard mode flag DF is set to 1 in S130. The DMG value determined in S127 becomes the next DMG value shown in Figure 33B.

The processes at arbitration timing T=3 are similar to those at T=2. When 80 bits are requested at T=4, the GAP value in S108 becomes 5. After going through the processes in S109 and S110, GAP and DMG are compared in S111. In this case, since GAP is equal to the preceding DMG, after going through the process in S112, V_i is set to 30 bits in S113, before going to the process in S116 shown in Figure 31.

In the processes from S118 to S124 shown in Figure 31, since V_i is already 30 bits, no more discarding is performed. After going through the processes in S125 and S126, the DMG is set to 4 in S127, DF is set to 1 in S130, and all processes for the channels are completed.

Although the number of requested bits is always 80 at and after T=5, the allocated bits are only incremented by 10 up to T=8. Finally, at T=9, the requested bits match the allocated bits. That is, in contrast to the silence sections at T=2 and 3, the period T=4 through 8 drag the discarding history, during which period the DMG value is decremented by 1. The period of dragging the discarding history ends when the DMG value becomes 0. That is, the variable DMG informs the next frames of the frequency of past discards.

Figure 33C illustrates dragging the history of discarding because of congestion. As in the case shown in Figure 33B, requested bits are allocated "as is" at T=1. However, after the GAP value is set to 5 in response to the 80 bit request in S108 shown in Figure 32, no bits are discarded, as shown in Figure 32. It is assumed that the processes S118 through S124 shown in Figure 31 cause 40 bits to be discarded because of congestion, so that the the number of allocated bits becomes 40. The DMG value at this time, which is the final value, is 4 after adding 4 to the preceding DMG value of 0 for the 40 bits discarded at S123. The DMG value of 3 after decrementing by 1 in S127 becomes the succeeding DMG value.

After T=3, the number of requested bits is always 80. Yet, at T=3, 4 and 5, since the succeeding DMG values are transmitted at the next timings after being decremented by 1, the number of allocated bits is incremented by 10 at T=3, 4 and 5, so that the number of requested bits matches the number of allocated bits at T=6.

Figure 34 shows an example of how respective variables in the flowchart shown in Figure 31 are determined in response to the preceding DMG value and the requested bits. In Figure 34, only the columnar contents have meanings, and the row arrangements are entirely irrelevant.

In the leftmost case, the number of requested bits is 40 and the preceding DMG value is 0. GAP becomes 1 in S108 in Figure 32. Assuming that all discard mode flags DF are 1 in Figure 34, GAP is compared with DMG in S111. Since GAP is greater than DMG, the DMG value does not change, but remains 0. This value is the DMG value shown under GAP in Figure 34. The value V_i does not change, and the process in S116 shown in Figure 31 is invoked. The value V_i at this time is the new V_i.

Thereafter, discards because of congestions are performed in S118 through S124 in Figure 31. The number of allocable bits for the channels in the discards because of congestion is 40. The value is expressed as the actual V_i. Hence, the requested bits "as is" become the final V_i, i.e. allocated bits, without being discarded, the final DMG value, the succeeding DMG value and the DF are all set to 0, and the processes are terminated.

The second column from the left shows a case in which the number of requested bits is 40, but the

preceding DMG value is 2. In this case, as shown in Figure 32, GAP is set to 1 in S108. Since DMG is judged to be greater than GAP in S111, DMG is set to 1 in S112, the V_i, i.e. the new V_i is set to 30 bits in S113, and the process in S116 is invoked. As described earlier, the number of allocable bits for congestion in this channel is assumed to be 40 in process S118 through S124. However, since the V_i is already 30 bits, its value becomes the final V_i and the process in and after S125 is invoked. Then, the DMG value is set to 0 in S127, which becomes the succeeding DMG, and the DF value becomes 0.

Since the preceding DMG is 2 when the number of requested bits is 70 in the fifth column from the left, after the new value V_i becomes 50 bits in S113 as shown in Figure 32, discarding because of congestion is performed in the processes after S118 shown in Figure 31. As described earlier, since only 40 bits can be allocated to the channel, 10 bits are discarded in S122, and DMG, i.e. the final DMG, is set to 3 in S123. Then, the DMG, value is set to 2 in S127, the resultant DF becomes 1, and the process is terminated.

The band arbitration processes explained in Figures 29 through 34 have the problem of determining how the past discarding history can be dragged. The mode of dragging the discarding history is set by a part of the content of the multiplexing parameters set in respective multiplexers 64 by the node controller 67 in Figure 21.

The following three modes can be considered for dragging the discarding history. First, there is the arbitrating mode shown in Figure 29, in which the discarding history is not actually dragged. Second, there is the arbitrating mode shown in Figures 31A and 31B, in which the discarding history is dragged and the restoration is performed stage by stage for each arbitration. Third, there is the arbitrating mode in which one stage is restored for every five arbitrations by decrementing DMGi by one for every five arbitrations in S127 shown in Figure 31B.

Figure 35 is a block diagram of an embodiment of a packet exchange network according to the high efficiency digitally multiplexed transmission method of this invention. In Figure 35, the multiplexer has exactly the same configuration as that shown in Figure 16A. A packet interface (PAD) 140 is inserted between a multiplexer 135 and a packet network 137a. Another packet interface (PAD) 140 is inserted between a multiplexer 136 and a packet network 137a. A packet interface (PAD) 140 assembles and dissembles packets, thereby transforming between multiplexed frames and packets.

Figure 36 is a block diagram of the embodiment of the packet interface 140 shown in Figure 35. In Figure 36, a header attacher 141 attaches a packet header to the packets received from the multiplexer side. After being assembled, packets are outputted to the packet network side through a speed difference absorption buffer 142. An arrival deviation absorption buffer 143 absorbs the deviations caused by packet exchanges of the packets received from the packet network side. A header filter 144 filters the packet header out. The packets are then transformed to multiplexed frames and outputted to multiplexers.

Figure 37 illustrates the operations of the speed difference absorption buffer 142 shown in Figure 36. In Figure 36, since the header attacher 141 attaches headers to the multiplexed frames inputted at speed V_1 from the multiplexer side, transmission speed V_2 to the packet network side is not the same as V_1. When they are transmitted at a uniform speed, V_2 becomes larger than V_1. The speed difference absorption buffer 142 absorbs this difference. When packets are transmitted in a burst to the packet network side when enough packets are accumulated and they are ready for transmission, the speed difference absorption buffer 142 is used as a transmission holding buffer. Figure 37 shows transmission mode at a uniform speed.

Figure 38 also illustrates the actions of the arrival deviation absorption buffer 143 shown in Figure 36. Even if the transmission side outputs packets to the packet network at a constant speed, the delays in the packet arrivals at the receiving side are not constant but have deviations, which the arrival deviation absorption buffer 143 absorbs, so that it outputs packets to the header filter 144 at a constant speed.

Figures 39A and 39B show an embodiment of a packet formatting. In Figure 39A, respective multiplexing frames with headers added to them become packets "as is" and are outputted to the packet network side. In Figure 39B, respective multiplexed frames are split into a plurality of packets, and respective packets have packet headers added to them, into which identifiers for reassembling multiplexed frames, etc. are inserted before they are outputted to the packet network side.

In Figure 39A, the multiplexed frame cycle is 5ms, the multiplexed frame length is 48 bytes, the execution throughput to the packet network side is 76.8kbps, and packets are formatted as ATM cells having 5-byte headers and 48-byte message data, for example.

In Figure 39B, the multiplexed frame cycle is 5ms, the multiplexed frame length is 96 bytes, the execution throughput to the packet network side is 153.6kbps, and packets are formatted as ATM cells having 5-byte headers and 48-byte message data, for example.

Here, although multiplexed frames are transmitted in split forms, since the multiplexers have a function of synchronizing multiplexed frames, the packet interface on the receiving side does not need to assemble frames, and there is no need to add extra control bits to the message data.

Figure 40 shows an embodiment of a code table for transmission band information at a band divisional coding illustrated in Figure 12. The transmission band codes for respective channels have e.g. 4 bits. Their three (3) least significant bits RI2, RI1 and RI0 are transmitted between the multiplexers (MUX) on the sending side and the demultiplexers (DMUX) on the receiving side. The most significant bit is used for notifying the discarding mode described later.

As shown in Figure 12, aside from the information of the respective bands having split side information, the numbers of transmission bits at respective discarding levels for bands B1 through B4 are shown as the numbers in the parentheses. For instance, the numbers of transmission bits for bands B1 through B4, e.g. at discarding level 1, are all 5. The transmission level at this time is 40kbps, and the transmission band information code is 110. At discarding level 5, the numbers of transmission bits for bands B1, B2 and B3 are 2 and the number of transmission bits for band B4 is 0, while the transmission level is 12kbps and the code is 010.

Figure 41 is a flowchart showing an embodiment of a multiplexation performed by a multiplexer for sub-band coding. In Figure 41, the minimum bands V_{min} for respective voice channels are set in S170. The minimum bands are different by channel numbers j. To reflect the past discarding history, the minimum bands for the channels changed from a silence mode to a speech mode and the channels previously set at discarding level 5 are set at discarding level 5, while the minimum bands for the channels previously set at discarding levels 1 through 4 are set at discarding level 4.

Next, in S171 to calculate the bands requested for voices, the sum of the transmission request bands is calculated for respective channels, excluding parts unnecessary for transmission for some voice channels among 50 supplementary bits, e.g. as shown in Figure 6. The remaining bands in the multiplexed frames are calculated in S172. Existences of remaining bands are judged in S173. If there is a remaining band (including 0), since the requested bands for respective channels can be multiplexed, the processes are terminated.

If it is not judged that there is any remaining band in S173, existences of discardable channels are judged in S174. If there is no discardable channel, the minimum bands for the respective channels are lowered to discarding level 5 in S175. If there are discardable channels, the process goes on to S176, skipping S175. The discarding level for the discardable channels is incremented by one level in S176, and the process reverts to S171. Then, the process continues until it is judged there is a remaining band in S173 when the process ends.

Figures 42A and 42B show an embodiment of a discarding mode corresponding to the voice frequency characteristics of the speaker. For example, in Figure 12, coders 393 through 396 for bands B1 through B4 measure voice powers for a predetermined period of time and calculate their average values. Figure 42A shows an example of the spectral distribution of the voice powers in the respective bands of a male speaker. The distribution is normalized by using the power in band B1. Figure 42B shows an example of a spectral distribution of the voice powers in the bands of a female speaker. Respective bands split in the spectral distribution by a threshold (indicated by a dashed line) are judged by their importance. In Figure 42A, discarding mode 1, emphasizing band B1, is used. In Figure 42B, discarding mode 2, emphasizing bands B1 and B2, is used.

Figure 43 shows an embodiment of the code table of transmission band information when a discarding mode is specified. In Figure 43, one bit of side information for identifying the discarding mode is used at a speech time in addition to the earlier described side information, together with an additional 1kbps for transmission level.

In Figure 43, at discarding level 5, mode 1, i.e. the number of transmission bits for a male speaker in respective bands B1 through B4, is the same as mode 2. At discarding level 4, unlike mode 2, mode 1 has 4 transmission bits for band B1 and 0 transmission bits for band B4.

Figures 44A and 44B are flowcharts of an embodiment for notifying the receiving side of a mode and for determining a discarding mode. Figure 44A is a flowchart of the case in which the sending side notifies the receiving side of a discarding mode, the receiving side returns the change receipt, and a discarding per the new mode begins. A voice coder notifies the multiplexer of the discarding mode predetermined as a default value before measuring the voice powers of respective bands in S178. The multiplexer notifies the demultiplexer on the receiving side of the default discarding mode in S179.

The coder on the sending side measures powers of respective bands in speech sections in S180, the coder judges the discarding mode per the speaker type in S181, and the coder notifies the multiplexer of the discarding mode in S182.

The multiplexer notifies the demultiplexer on the receiving side of the discarding mode in S183. The demultiplexer on the receiving side notifies the multiplexer of the receipt of a change in S184. The multiplexer notifies the demultiplexer on the receiving side of the beginning of the change, and a discarding

per the new mode begins.

Figure 44B is a flowchart of the process for notifying the receiving side of a discarding mode as side information of a voice. In this case, a discarding mode is changed, even though the demultiplexer on the receiving side does not notify the multiplexer of the receipt of a change. The coder notifies the demultiplexer of a default mode via the multiplexer in S186. The coder measures the powers in respective bands in S187, determines the discarding mode in S188, notifies the demultiplexer on the receiving side of the discarding mode as the side information of a voice via the multiplexer in S189.

The methods for determining discarding modes, illustrated in Figures 42A, 42B, 43, 44A and 44B, are explained in further detail below. The powers of the respective bands in speech sections are measured as described below. The processes are equivalent to those in S180 in Figure 44A or S187 in Figure 44B.

n samples of the powers of respective bands are measured at cycle intervals for judging speech and silence sections, e.g. a few milliseconds to a few tens of milliseconds. The powers of the respective bands at the intervals of judging cycles are expressed below as the input signal components of the band objected to S_k.

$$P(i) = \sum_{k=ixn}^{ixn+(i+1)xn-1} S_k^2$$

where i is incremented by n.

Next, the electrical power values calculated from the above equations are accumulated only when there is a speech in a judging cycle for obtaining the average power of respective bands, e.g. in 1,000 cycle periods for judging speech or silence. The accumulated value is divided by the number of accumulated terms. That is, the average electric power for the speech period is obtained from the following equation.

$$\overline{P} = \left(\sum_{k=1}^{1000} VDF_i \times P(i)\right) / \left(\sum_{k=1}^{1000} VDF_i\right)$$

where VDF_i is a speech flag, becoming "1" when speech is judged to exist.

Next, the determination of the discarding mode in S181 or S188 is based on the calculation of the frequency characteristic distributions, as illustrated in Figures 42A and 42B. The spectral distribution for band B_2 is obtained by dividing the average electric power of B_2 by assuming the average electric power during a speech period for band B_1 is 1. Figure 42 illustrates the method of judging the spectral distribution of all bands based on a constant threshold. Yet, it is possible to change the threshold for each band in an actual use.

Figure 45 is a block diagram showing an embodiment of a compressed transmission system regarding the channels in a holding mode as being in a silence mode. In Figure 45, when the line is held, the transmission band is compressed as a silence mode without transmitting a holding sound. The sending side notifies the receiving side of the holding mode by inserting the information e.g. in the R_i code described earlier.

As shown in Figure 45, the holding mode detector (R-DET) 191 on the sending side detects a line holding directly [1] when a holding button of the telephone is depressed, [2] when a PBX detects a holding sound pattern outputted from a telephone unit, or [3] when the input side of the voice coder detects a holding sound pattern. The holding sound mixer (R-MIX) 192 on the receiving side mixes holding sounds [4] on the output side of a voice coder, [5] in a PBX, or [6] on another telephone unit. Clearances of holding mode is notified to the R-DET 191 [1] through a change in a hooking condition of a telephone or clearing the holding button, [2] through a mismatch of a holding sound on the input side of a coder, or [3] through a change in the hooking condition on a PBX, with a combined use of a monitor of a signaling signal. Thereafter, voices are transmitted to the receiving side per the earlier described method.

A holding mode detector (R-DET) such as one shown in Figure 45 can be contained in the block of a PBX, a COD (coder) or MUX (multiplexer). Figures 46A and 46B show embodiments for containing a holding mode detector in a switching system. In Figure 46A, the holding mode detector R-DET provided in a switching system outputs a holding sound detecting signal, when it detects a holding mode.

Figure 46B illustrates a method for judging the holding mode. According to this method, the holding mode is judged to be in effect when the receiving side is in an on-hooked mode for a predetermined time period during a call connection, and the holding mode is judged to be cleared when the receiving side is in an on-hooked mode for a predetermined time period during a holding. Such predetermined time periods

can be anywhere between 50ms and 1 second, for example. Oftentimes, the call is disconnected when the calling side is hooked on during a call connection, although the called side cannot disconnect the call unless the on-hooked mode is maintained for about 30 seconds. Therefore, such a holding becomes possible.

Figure 47 shows an embodiment of a holding mode detector R-DET contained in a multiplexer. In Figure 47, the holding mode detector R-DET receives a signaling signal from the exchanger and detects the holding mode by a method similar to that shown in Figure 46B.

Figure 48 is a block diagram of the schematic configuration of the system in which multiplexers accommodate various transmission media. In Figure 48, the modem connected to a terminal in addition to a telephone, a telephone and a fax machine are connected to a PBX. The multiplexer (MUX) in a multiplexing device comprising coders (COD), a multiplexer (MUX) and a network port (NP)] transmits its output to the transmission path through the network port (NP). In this case, it is not possible to ascertain which fax machine prohibited from discarding information is connected to which coder through the PBX.

That is, in Figure 48, since some transmission media set all information equivalent to the core information and do not have any supplementary information, the coder for coding the information sent from such media need to code the requested band of the transmission media without discarding any of them. Therefore, there is a need to detect e.g. the fax protocol, thereby enabling the coder receiving the detection signal to code input information without discarding it.

Figures 49A, 49B and 49C are block diagrams showing an embodiment of the coder coding per the fax protocol detection result. Figure 49A is an example of a coder sending a coded signal to a multiplexer without discarding any requested band, on receipt of a detection signal from a FAX protocol detector not shown in the drawings.

Figure 49B shows an embodiment of a coding when the coder itself detects a FAX protocol, where input information is coded without discarding the requested band when the input signal is a fax protocol, and where the input information is coded according to the earlier described discarding method when the input signal is a voice. Figure 49C shows an example in which both an ordinary voice coder and a fax protocol terminator receive input signals split at a junction and the multiplexer receives the output from the fax protocol terminator through the selector when a fax protocol is detected or the output from the ordinary voice coder when a fax protocol is not detected.

Figure 49B shows a method for a coder to detect fax protocols. The detection method is explained by using the outline of transmission sequence of G3-FAX shown in Figure 50. In Figure 49B, a coder judges fax data when it detects a transmission of a CNG tone or a CED tone at phase A.

Although Figure 48 illustrates a system for accommodating various transmission media, it is conceivable to change the coding method used by the coder per the medium characteristics. For instance, in Figure 49B, the coder can detect a fax protocol as described earlier thereby selecting a band in correspondence with a fax signal. In this case, a coder using a fax protocol terminator in combination, as shown in Figure 49C, need not be used.

For a touch-tone telephone, if a coding such as for sending only a tone code when a tone signal is detected, transmissions can be made at a very low rate such as below 1kbps. Furthermore, signals such as those in personal computer communications can be coded in bands different from those for ordinary voice conversation, and MODEMs using different modulation methods can differentiate their necessary coding speeds.

A system such as that shown in Figure 48 needs to limit the number of channels not allowing discarding any parts of the requested bands and to prevent at least such channels not allowing discarding from actually discarding channels. A method based on number programming, for example, is used to limit the number of channels. In Figure 48, when the input port on the multiplexing complex side for accommodating a fax is limited to #1, the fax dial numbers are programmed to be e.g. 7xxx, and ordinary phone numbers are programmed to be e.g. 8xxx. By registering such a numbering system in the PBX, connecting the call to port #1 when the dial numbers are 7xxx and to ports #2 through #n when the dial numbers are 8xxx, even if two fax lines are to be extended, since port #1 is busy, the second call is blocked and the number of channels not allowing discarding is limited.

When channels not allowing discarding exist, the channels allowing discarding need to further increase the number of discardable bits. For instance, in Figure 12, although core bits are transmitted for bands B1, B2 and B3 at level 5, it is necessary to allow discarding until level 6 at which no core bits are transmitted, i.e. a silence mode.

Figure 51 shows an embodiment of the processes added to the flowchart shown as Figure 41. More specifically the processes shown in Figure 51 are to be inserted between S175 and S176 in Figure 41. After the minimum bands for the respective channels are set at level 5, it is judged whether or not there is any

discardable channel in S200. When there is no discardable channel, the minimum band for the channel allowing discarding like DSI until a silence mode is set at level 6 in S201 before the process goes on to S176. It goes without saying that the process in S176 is performed, skipping processes in S200 and S201, when there is a channel allowing discarding in S174 in Figure 41.

Figure 52 is a block diagram of the system for halting the operation of the voice detector or ignoring the detection result, when the number of voice channels to which call connections are performed are small, when there is no need to discard the supplementary information part, or when the sound deterioration caused by the discarding of the supplementary information part is less than that caused by the voice detector. In Figure 52, e.g. the multiplexer (MUX) counts the number of calls to determine the necessity of discarding, and prevents a sound quality deterioration caused by a voice detector by outputting a voice detector ignoring/halting signals to a coder when there is no need for discarding.

In this manner, when the number of calls is small and there is no possibility of discarding information, halting voice detector operations can prevent sound quality from degrading. The voice detector is used for detecting a speech or silence mode and for improving the transmission efficiency within the network by cutting unnecessary voice information, although it naturally causes a sound quality deterioration compared with the case in which a voice detector is not used. For instance, the microphone of a telephone hand-set picks up both the speaker's voice and background noise. If the voice information is cut as the silence mode when the speaker does not utter a sound, the atmosphere is not transmitted, because the background noise is not transmitted. Although the purpose will be served if the silence mode is detected by judging the content of the conversation, an ordinary silence mode detector sometimes judges even the part that should not be cut as a silence mode to be eliminated, which causes a sound quality deterioration. Since a sound quality deterioration can occur similarly even when the number of used calls is small and nothing is discarded, by mutating the operation of the voice detector, the sound quality can be improved.

When relay switching system are provided between the switching system or the multiplexer of the information originator and the switching system or the multiplexer of the information receptor, it is possible to improve the sound quality of the transmitted information, if the switching system or the multiplexer of the information originator collects data on the numbers of channels to which calls are connected in the switches of the respective relay stages by way of the communication paths for the direction opposite to the information transmitting direction and the voice detector of the information originator ceases operating or ignores the detecting result, when the supplementary information part needs not be discarded. Since the number of voice channels to be connected is selected by assuming the most congested mode in a designing ordinary line, the possibility that something actually needs to be discarded is remote, and the sound quality can be improved even when the switchings are repeated in multiple stages. In this case, it is possible to achieve the purpose by having the switching system of a relay stage to which fewer channels are call-connected transmit a signal instructing to cease the operations to the voice detector of the information originator by way of the communication path for the direction opposite to that in which the information is transmitted, instead of having the switching system or the multiplexer of the information originator collect data on the numbers of channels of the switches in respective relay stages.

Next, a method for judging whether or not the Sound quality deterioration caused by discarding the code information is less than that caused by the operation of the voice detector is explained by referring to Figure 52, which is judged simply by the relations among the line capacity, the degree of multiplexation, and the number of channels actually carrying calls. For instance, assuming that an evaluation was already made for the coder having the characteristics shown in Figure 23, the sound quality is better if the silence mode detector is mutated for up to 5 channels when a 64 kbps line is multiplexed. This means that a result is known that the deterioration caused by coding is less than that caused by the transmission of a silence mode in which no code information is sent, even if some bands like 11kbps are discarded in a stage-by-stage discarding when all channels carry sounds.

This case is equivalent to a case where, because of an overhead of the headers in multiplexed frames, no supplementary information part is discarded e.g. for at least up to three channels and the sound quality deterioration caused by discarding code information is less than that caused by the operation of the voice detector in a multiplexation of 4 or 5 channels.

Figure 53 is a block diagram showing the basic configuration of a pipeline multiplexing method for multiplexed transmissions of a plurality of multiplexed data split from the multiplexed data in one multiplexing unit in line sets. If a multiplexer (MUX) performs a multiplexation after receiving all data in one multiplexing unit from the line sets, the lag before the multiplexed data are outputted to a transmission path becomes large. Also, since the demultiplexer (DMUX) demultiplexes the multiplexed data in one multiplexing unit, the delay is further exacerbated.

Ordinarily, about 10ms is appropriate for one multiplexing unit, but the effects of the delays become

larger when relays are performed in multiple stages.

Therefore, a pipeline multiplexation method is devised for reducing the delays by further splitting the data in one multiplexing unit. This method is illustrated in Figure 53.

In Figure 53, the split number for further splitting one multiplexing unit in line sets 210a, 210b, ... is assumed to be n in explaining the pipeline multiplexing method.

In Figure 53, line sets 210a, 210b, ... output data in one multiplexing unit in a form that can be split into n, according to the split number n. A multiplexer 211 in a multiplexing unit receives information indicating the transmission request band (or information indicating the existence of a need for transmission) in the data in one multiplexing unit prior to the target multiplexed information or contemporaneously with its head end.

The multiplexer 211 in the multiplexing unit comprises a band arbitrator 213 for arbitrating the bands during a discarding for congestion and a pipeline multiplexer 214 for pipeline multiplexing data in one multiplexing unit from line sets 210a, 210b, ... per the arbitration result. The multiplexation and the output to the transmission paths for the respective spit data are consummated in 1/n of the processing times required for data in one multiplexing unit from line sets 210a, 210b, ... When the multiplexed data in the first split are received, bands are arbitrated per the information on the transmission request band already or contemporaneously received, so that all pipeline multiplexations are consummated in 1/n of the time required for the multiplexation of data in one multiplexing unit.

Next, the demultiplexer 212 in the multiplexing unit comprises a batch demultiplexer 215 for demultiplexing the respective channel data in one multiplexing unit and outputting them to line sets 210a, 210b, ... and a split demultiplexer 216 for sequentially demultiplexing the multiplexed data starting from the first split and for outputting to line sets 210a, 210b, ... immediately after the demultiplexation.

Per the pipeline multiplexation method illustrated in Figure 53, of the multiplexing information arriving from the transmission path only that necessary for restoring the information indicating the transmission bands of respective channels are first received. Per the restored information the pipeline multiplexed data are split demultiplexed and outputted to line sets 210a, 210b,

That is, in Figure 53, the multiplexer 211 sequentially multiplexes data upon receiving 1/n of the data of one multiplexing unit and outputs them to the transmission path. Therefore, delays caused by multiplexation are less than when data are multiplexed after all in one multiplexing unit are received. Also, since the demultiplexer 212 sequentially demultiplexes data in one multiplexing unit through split demultiplexation, the delays caused by demultiplexation are less than when data are demultiplexed in a batch.

Figure 54 shows an embodiment of the pipeline multiplexing method. Figure 54 illustrates a case where $n=2$ for two channels, i.e. where the data in one multiplexing unit for two channels are are split into two.

Data [2] and [4] in one multiplexing unit (10ms) for both channels are respectively set at 40 bits. Data [1] and [3] indicating the transmission request bands for both channels are respectively set at 2 bits. The transmission capacity of the transmission path, i.e. the length of the fixed length frame [5] is set at 64 bits.

First, the multiplexer 211 receives data [1] and [3] indicating the transmission request bands for arbitrating the bands. Since the sum of data [1] through [4] is 84 bits, and a band of 20 bits is deficient, the bands are arbitrated. The results of this sarbitration are as shown in Figure 54B.

Second, the multiplexer 211 receives first split data [2]-1 and [4]-1 and multiplexes the first split data for channel 2 per arbitration results [a] and [b]. For notifying the receiving side of the occurrence of discarding at multiplexations, transmission band data [c] and [d] are split and attached. As a result, the first pipeline multiplexation data, shown in detail as [5]-1 in Figure 49C, are sent to the transmission path.

Third, the multiplexer 211 receives the second split data [2]-2 and [4]-2 to perform similar multiplexations per the arbitration results. The multiplexed data, shown in detail as [5]-2 in Figure 54C, are sent to the transmission path.

The split demultiplexation by the demultiplexer 212 is explained next by referring to Figure 54. The demultiplexer 212 receives two split demultiplexed data over a time period for one multiplexing unit, from which data indicating transmission bands are restored, which identify which of the received multiplexed data are to be assigned to which channels by how much.

Per transmission band data [c] and [d], 0 bits and 20 bits are demultiplexed from first pipeline multiplexed [5]'-1 respectively for channels 1 and 2. They are sent to the line sets of respective channels together with the restored transmission band data. Then, 0 bits and 20 bits are demultiplexed from second pipeline multiplexed [5]'-2 for channels 1 and 2, respectively. These bits are then sent to the line sets of the respective channels.

Figures 55A and 55B show an embodiment of the pipeline multiplexation in one multiplexing unit. Figure 55A shows multiplexed data Xi in one multiplexing unit for channel i and transmission request band data Ri therefor. Data in one multiplexing unit are split equally into 1/n, and the split data are named Xi1,

Xi2, ... and Xin corresponding to their arrival sequences. Transmission request band data Ri are also equally split by n to be named Ri1, Ri2, ... and Rin. Since some parts of these split data are discarded as a result of the band arbitration, the data actually transmitted become Xi1', Xi2', ... and Xin'.

Figure 55B is a flowchart of the pipeline multiplexation. In Figure 55B, a band arbitration is performed in S218 by any one of the earlier described arbitration methods. Then, the numbers of bits Xi' allocated for transmission are determined for the respective split data to be actually transmitted to the respective channels.

Then, in S219, n is set to 1. In S220, the first split data for n pipeline multiplexations are multiplexed. Here, the first split data for all channels are received, and the split data allocated for transmission are extracted. Transmission band data Rin and allocation data Xin' are multiplexed over the regions having 1/n of the transmission capacity C of the transmission path to be outputted to the transmission path, starting from channel 1. In S221, the n value is incremented, and the process in S220 is repeated until the n-th one of the split data.

Figure 56 is a flowchart of the batch demultiplexation performed by the demultiplexer 212 shown in Figure 54. In S222, n indicating the sequence of the split data is set to 1. In S223, the first split data are stored. In S224, the n value is incremented. The process in S223 is repeated until the n pipeline multiplexed data are stored.

After N split data are stored, transmission band data Ri for respective channels are restored from data for one multiplexing unit stored in S225. In S226, i indicating the channel number is set to 1. In S227, split data transmitted to the channel i per the value of transmission band data Ri are extracted to be combined in accordance with the sequence. In S228, data Xi are sent to the line sets for channels i. The processes in S227 and S228 are repeated by the number of channels until the batch demultiplexation is completed.

Figure 57 is a flowchart of an embodiment of split demultiplexation. In Figure 57, the processes up to S225 are exactly the same as those of the batch demultiplexation shown in Figure 56.

In S229, n, indicating the sequence of split data, is set to 1 . In S230, i, indicating the channel number, is set to 1. In S231, transmission data Xin' for channel i are extracted from the stored n-th multiplexed data per the value of transmission band data Ri. In S232, the data are outputted to line sets for channels i. After repeating the processes in S231 and S232 for the number of channels, the processes in S230, S231 and S232 are repeated for the number of split data n, and the split demultiplexation is completed.

Figure 58 illustrate the delay reduction through the pipeline multiplexation method. In Figure 58, the time equivalent to one multiplexing unit in line sets (basic multiplexing unit) is set to t, and is compared with the lag time in n pipeline multiplexation. Through n pipeline multiplexations, the multiplexer causes only 1/n lags in receiving line set data, 1/n delays in multiplexings and 1/n delays in outputting multiplexed data to the transmission path. Therefore, the processing time for one multiplexing unit becomes virtually t/n, and the multiplexing lags are effectively reduced. The n pipeline multiplexations cause the lag time to be reduced from 2t to $(n+1)t/n$.

Consequently, since the total delay time reduction T is expressed as: T = (lags by non-pipeline multiplexers + lags by non-pipeline demultiplexers) - (lags by n pipeline multiplexers + lags by n pipeline demultiplexers) = (2t + 2t) (2t/n + t + t/n) = 3t -3t/n Where n is large, lags equivalent to three times those for one multiplexing unit are reduced.

Figures 59A and 59B show methods for protecting the voice quality by simultaneously notifying the coder and the multiplexer of the voice detection information.

Figure 59A shows a conventional method for protecting voice quality by simultaneously notifying the coder and the multiplexer of the information on voice detection. A voice detector outputs information on voice detection regarding whether or not there is a sound during a predetermined time period. Naturally, the output of the judging result lags behind the output of the voice data by the predetermined time period. Therefore, the information on voice detection is sent to the multiplexer after its phase is matched with that of the code information processed by the voice coder.

The problem in detecting a voice lies in the change point from a silence mode to a speech mode such as at the beginning of a sentence or at the beginning of a word. Although various methods can be considered for the detecting such change points, a guard time method for protecting the speech mode by outputting the information on the voice detection essentially at an earlier point for producing more naturally reproduced sounds becomes necessary. However, as described earlier, since the guard time method causes the information on the voice detection and the code information to be sent to a voice coder by matching their phases, voice information needs to be stored in the memory, first. This not only requires extra memory capacity but also causes larger delays in voice transmission.

Figure 59B shows the method of this invention for protecting voice quality by simultaneously notifying the coder and the multiplexer of the detection result by the voice detector. Hence, the detection result is

notified to the multiplexer in advance of the time required for the processing by the voice coder, i.e. the time required for coding. This enables the beginning of a sentence or the beginning of a word, i.e. the points where a silence mode changes to a speech mode, to be protected, thereby assuring the protection of the voice quality without causing an additional delay.

Figure 60 shows an embodiment of the method for generating a noise on the receiving side in correspondence with the noise level of the sending side in a silence section. Generally, no voice data are transmitted from the sending side to the receiving side during a silence mode, and the receiving side reduces the users' sense of-severance or strangeness at the time of switching from a silence mode to a speech mode by inserting noises to the silence mode. Usually, the noise level inserted on the receiving side is set at a constant level, irrespective of the noise levels on the sending side. This causes a problem that a sense of awarkwardness is aroused when the noise level on the sending side differs greatly from that on the receiving side.

To overcome such problems, this invention causes the noise level to be measured simultaneously, when a voice detector on the sending side detects a silence mode, and the information indicating the level to be sent to the receiving side, as shown in Figure 60A. Also, this invention causes the receiving side to create noises for the silence section based on the noise level information, as shown in Figure 60B.

As described earlier, when signals for signaling voice channels are transmitted by signal-sent (SS) signals and signal-received (SR) signals, if the logical sum of SS and SR signals remain "0" for a predetermined period of time, a call connection mode is judged to exist. If their logical products remain "1" for a predetermined period of time, a call severance mode is judged to exist.

Either the coder or the multiplexer can detect the calls, because signaling (SS and SR) signals are transmitted through the voice coder and the multiplexing unit. When a call detection is performed through a software processing, only the algorithm shown in Figure 9C needs to be used. This algorithm is executed by a processor 77 in the multiplexing unit shown in Figure 26. When a call detection is performed through a hardware processing, only call detection circuits described later need to be provided in the respective voice coders, or alternatively in the multiplexing unit for the number of voice channels.

Figure 61 is a block diagram of an embodiment of a call detection circuit. The call detection circuit shown in Figure 61 comprises an OR gate 235 for receiving both SS and SR signals, a delayed type flip-flop 236 for receiving at its delayed input terminal the outputs from the OR gate 235, a counter 237 for receiving at its reset terminal the output Q from the delayed type flip-flop 236, a counter 238 for receiving at its reset terminal the output $\overline{Q}$ from the delayed type flip-flop 236, and an RS flip-flop 239 for receiving respectively at its set terminal and reset terminal the carry outputs from the two counters 237 and 238.

In Figure 61, when both SS and SR signals are 0, the output Q from the delayed type flip-flop 236 is 0 and the output $\overline{Q}$ from the delayed type flip-flop 236 is 1. Then, the counter 238 is reset, and the counter 237 is counted up at each receipt of a clock pulse and outputs a carry signal after a predetermined time, which is $TAU_1 = 640ms$ in this case. The RS flip-flop 239 outputs Q = 1 as the detection result of the call connection mode.

On the other hand, when both SS and SR signals are 1, the counter 237 is reset, the counter 238 counts the clock pulses until $TAU_2 = 640ms$ at which time it outputs a carry signal, the RS flip-flop 239 outputs 0 as the output Q indicating a call severance.

A system transmitted signaling information as SS and SR signals utilizes the character that signaling signals hardly change except for the period when the call origination is processed so as to reduce the necessary bands when signaling is transmitted per the out-band method. Figure 62 shows an example of signaling signals. In Figure 62, after it is activated, the calling side keeps the constant value of 0 until the start of the selection numeric signal transmission, has a value of either 0 or 1 when a selection numeric signal is being transmitted, and has a constant value of 0 again thereafter. The receiving side has the value of 1 until it responds, but has the unchanged value of 0 after responding.

This invention invokes a transmission compression only when the signaling bits, e.g. 4 bits, of the frames transmitted immediately before are all 0 or all 1, and the signaling bits of the frames currently transmitted do not change at all. The reason why the condition of all 0 or all 1 is added is to ensure the tolerance for the line random error.

That is, the receiving side processes elongation at a compression based on the all-0-or-all-1 rule before proceeding on the transmission compression, and corrects an error arising on the transmission path by the majority logic, when it actually happens. In addition, media other than voices, such as remote signaling (RS) signals for data channels, i.e. level signals for fixing the mode between terminals, can use an entirely similar transmission compression method.

Figures 63A and 63B are flowcharts of an embodiment of the signaling compression. Figure 63A shows the processing flow on the sending side in the embodiment. After the process begins, it is judged whether

the signaling data are all 0 or all 1 in S240. If they are neither all 0 nor all 1, a signaling transmission flag is provided with 1 indicating a transmission of signaling information in S241 before terminating the process.

When the signaling data are either all 0 or all 1 in S240, it is judged in S242 whether or not the signaling data are changed from those for the preceding frames. When the signaling data, are changed, since signaling data need to be sent, 1 is provided in the signaling transmission flag in S241 before terminating the process. When the signaling data are not changed from those in the preceding frames, as a compression mode, a signaling transmission flag of 0 is set in S243 and the process is terminated.

Figure 63B shows an embodiment of the signaling compression on the receiving side. When the process begins, it is judged whether the signaling transmission flag received in S244 is 0 or 1. When the transmission flag is 1, since it means that the signaling data are transmitted, the received signaling data are retained in S245, and they are outputted in S246 to terminate the processing.

When the signaling transmission flag is 0 in S244, it is judged whether or not the signaling data currently retained as an elongation mode, i.e. the last received signaling data, are all 0 or all 1 in S247. When they are either all 0 or all 1, their retained values are outputted in S248, and the process is terminated.

Since there is a transmission error when they are neither all 0 nor all 1, it is judged in S249 whether the number of 1 is 1, 2 or 3. When it is either 1 or 3, the error is corrected based on the majority logic. When it is 2, it is considered as a line severance mode in S251 so that they are corrected to all 1, and the corrected retaining value is outputted in S248 and the process is consummated.

As explained so far, this invention causes the core information part for assuring the minimum acceptable sound quality for the voice channels to be necessarily multiplexed sequentially from the parts with the highest priority among the supplementary information part necessary for obtaining the desirable sound quality, thereby minimizing the deterioration in communication quality even when some bands heed to be discarded during a congestion time. Media capable of waiting, such as packet switching system multiplexed voice channels after the necessary bands arc secured for the packet data of the minimum packet switch throughput, thereby multiplexing the voices in the voice channels in the remaining bands and multiplexing packets in the residual bands. This enables efficient formation of multiplexed frames. Thus, this invention enables the concurrent pursuit of both communication quality improvement in statistical multiplexations and efficiency improvements in multiplexations.

This invention realizes a highly efficient, digitally multiplexing system for use as a multiplexing device using a plurality of voice communication lines over future digital communication lines such as broad-band ISDN, current ISDN, digital private line services such as "high bit link" international service, voice communication switching networks such as PBXs and public networks, and multi-media switching networks.

So far we have explored the possibility of applying this invention to efficient data transmission. However, the application of this invention is by no means limited to sound transmission only. It is applicable, for example, to image data transmission.

Figure 64 is a block diagram showing an embodiment of an image processing system to which this invention is applied. This embodiment is similar to the embodiment illustrated in Figure 5B. The same parts have the same numbers, and their detailed explanations are omitted here. The only difference is that an image coding means 20' is used in lieu of the voice coding means 20.

Figure 65 shows an example of determining the request band. At rate level 0 with transmission rate of 8kbps, only side information is transmitted. At rate levels 1, 2 and 3, the transmission rates are 256kbps, 1024kbps and 2048kbps, respectively. Based on rate levels, band arbitration is performed.

Claims

1. A high performance digitally multiplexed transmission system for use in a multiplexing device for transmitting multiplexed signals on a digital line, said high performance digitally multiplexed transmission system comprising:

a voice coding means for separating voice input information into a core information part for ensuring minimum acceptable sound quality and, in a supplementary information part, for sending supplementary information to be sent with said core information part for getting desirable sound quality, parts of said supplementary information part discardable in sequence according to the priorities, said voice coding means for coding said core information part and said supplementary information part;

a detecting means for detecting a silence section in said voice input information; and

a multiplexing means for multiplexing, among a plurality of voice channels to be multiplexed, only information necessary for synchronization with a correspondent's coder for a voice channel where said silence section detecting means detects a silence section, for multiplexing said core information part for

a fixed length digital slot repeated at a predetermined cycle for a channel where said silence section detecting means detects no silence sections, for multiplexing said supplementary information part of each voice channel in the order of higher priority where no silence sections are detected for the remaining part of said fixed length digital slot, and for discarding a supplementary information part which cannot be multiplexed due to insufficient band.

2. A high performance digitally multiplexed transmission system according to claim 1 comprising a request band determining means for determining each transmission request band for voice input information for a plurality of voice channels to be multiplexed in said digital line, wherein:

a supplementary information part within a request band determined by said request band determining means is discarded in the order of lower priority when a supplementary information part of each voice channel where said multiplexing means detects no silence sections is discarded.

3. A high performance digitally multiplexed transmission system according to claim 1 or 2 comprising a call detecting means for detecting a call in each of a plurality of said voice channels to be multiplexed, wherein:

said multiplexing means does not transmit any coding information including any information required for synchronization with said correspondent's coder specifically for a voice channel where said call detecting means detects a calling mode.

4. A high performance digitally multiplexed transmission system according to claim 1, 2, or 3 comprising a packet assembling means for assembling a packet using a multiplexed frame provided by said multiplexing means "as is" or in a divided form, wherein:

said multiplexing device applies said packet to the label multiplexing network connected to users' devices.

5. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 4 comprising a node control part, in said multiplexing device, for providing a multiplexing parameter based on the nature of sound quality deterioration due to the discard of said supplementary information part which depends on the characteristics of each coder for a plurality of voice channels to be multiplexed in said digital line, and for setting said multiplexing parameter in said multiplexing means, wherein:

said multiplexing means determines the method of discarding a supplementary information part in the current transmission frame according to said multiplexing parameter and the discard history of the supplementary information in the preceding transmission frame.

6. A high performance multiplexing transmission system according to claim 1, 2, 3, 4, or 5 in said multiplexing device that multiplexes transmission information of data channels, being other media, including data terminals in addition to a plurality of voice channels and transmits said multiplexing signals in a digital line, said system comprising:

a significant information detecting part for detecting the necessity of information transmission for each of said data channels, wherein:

said multiplexing means multiplexes for a plurality of said voice channels after multiplexing transmission information of data channels which said significant information detecting part considers necessary to be transmitted.

7. A high performance digital multiplexing system according to claim 1, 2, 3, 4, or 5 in said multiplexing device that multiplexes transmission information of data channels including data terminals and packet data from a packet switching system in addition to a plurality of voice channels and transmits said multiplexing signals in a digital line, said system comprising:

a significant information detecting part for detecting the necessity for information transmission for each of said data channels, wherein:

said multiplexing means reserves a minimum throughput guarantee value of said packet switching system, multiplexes transmission information of data channels which said significant information detecting part considers necessary to be transmitted and packet data of the minimum throughput in said packet switching system, multiplexes for a plurality of said voice channels, and multiplexes packet data in the remaining part of said fixed length digital slot.

8. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, 4, 5, 6, or 7 wherein:

said voice coding means comprises a sub-band coder; and

a multiplexing part forming said multiplexing process means selectively discards a core information part corresponding to the specific frequency component when said multiplexing part cannot transmit all core information parts for each of said voice channels where no silence sections are detected.

9. A high performance digitally multiplexed transmission system according to claim 8, wherein:

said voice coding means comprises a sub-band coder;

a multiplexing part forming said sub-band coder or said voice multiplexing means detects the voice frequency characteristics of each of said voice channels;

said multiplexing part performs a multiplexing process including selective discard of said core information part based on the discard mode corresponding to the detected frequency characteristics.

10. A high performance digital multiplexing system according to claim 1, 2, 3, 4, 5, 6, 7, or 8, wherein:

said multiplexing means reserves a code table for coding a transmission band, and codes the transmission information band for each channel in a transmission frame, said transmission information being actually transmitted through said digital line.

11. A high performance digital multiplexing system according to claim 1, 2, 3, or 4 in said multiplexing device which notifies a correspondent's multiplexing device of the pending mode of said channel when detecting a channel in the pending mode among a plurality of said voice channels, wherein:

said correspondent's multiplexing device outputs a built-in-pending-sound pattern and attempts to compress transmission when detecting the silence mode on a relay transmission line between said multiplexing devices.

12. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 5 in said multiplexing device which comprises various transmission media for transmitting voice, for example, wherein:

a coder forming said voice coding means encodes after arbitrating the number of bits in a core information part, for assuring information quality necessary for respective transmission media, where a coding method or discarding may not be admitted depending on the nature of each medium; and

a multiplexing part forming said multiplexing process part performs a multiplexing process corresponding to each of transmission media according to a request from each of said encoders.

13. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 4 for avoiding deterioration of sound quality by stopping the operation of a voice detector or ignoring the detecting result of said voice detector when said multiplexing device or switching system detects the number of channels where calls are connected among a plurality of voice channels and it is not necessary to discard said supplementary information part or when the sound quality deterioration due to the discard of said supplementary information part is less than that due to the operation of the voice detector forming said silence section detecting means.

14. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 4, wherein:

said multiplexing device or a switching system originating information collects data on the number of channels connected to calls in switching systems relaying said information in a label multiplexed network such as said packet network or ATM network via a communication path opposite to the direction for transmitting information; and

unnecessary sound quality deterioration can be avoided by stopping the operation of a voice detector forming said silence section detecting means when a discard of said supplementary information part is not required or by ignoring the detection result of said voice detector.

15. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 4, wherein:

line sets forming said voice coding means and silence section detecting means outputs one multiplexed unit of multiplexed data in said line sets and information indicating the necessity of transmission of said data or indicating a transmission request band;

said multiplexing process means first takes the information indicating the necessity of transmission of said data or indicating a transmission request band, and comprises a band arbitrating part for

determining a transmission band for each channel according to said information and a pipeline multiplexing part for obtaining data after dividing one multiplexed unit of multiplexed data from said line sets into a plurality of split data groups according to the result provided by said band arbitrating part and for multiplexing and transmitting said divided data groups every time information is obtained; and delay time can be greatly reduced in association with the multiplexing operation.

16. A high performance digitally multiplexed transmission system according to claim 15, for greatly reducing lags associated with demultiplexations, wherein:
said multiplexing process means first restores the information indicating the transmission band of each channel from one multiplexed unit of multiplexed data in said line sets received through the transmission line:
said high performance digitally multiplexed transmission system further comprising:
a pipeline demultiplexing part for demultiplexing data into groups in the order of their multiplexations according to said restored information and outputting them to said line sets.

17. A high performance digitally multiplexed transmission system according to claim 15 or 16, wherein:
said line sets comprises line sets for a data channel in addition to line sets forming said voice coding means and silence section detecting means.

18. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 4, wherein:
the determination result of detecting speech/silence sections by a voice detector forming said silence section detecting means is sent simultaneously to a coder forming said voice coding means and a multiplexing part forming said multiplexing process means; and
voice information quality can be protected without increasing delay time of voice information longer than the time required for voice coding by said coder.

19. A high performance digitally multiplexed transmission system according to claim 1, 2, 3, or 4, wherein:
a voice detector forming said silence section detecting means measures the noise level on the sending side for the voice channel where a silence section is detected;
said multiplexing process means multiplexes, in addition to the information required for synchronization with a correspondent's coder, the information indicating said noise level for said fixed length digital slot for said channel; and
noise is generated for a silence section on the receiving side according to the information indicating said noise level.

20. A call detecting method in a voice multiplexing device, a multi-media multiplexing device, and a switching system that transmits signaling information of a voice channel as a signal-send (SS) signal and a signal-receive (SR) signal, wherein:
said call detecting system detects the calling mode or non-calling-mode of said voice channel based on the time period showing a certain value of logical OR or logical product while monitoring said SS and SR signals and after obtaining the logical OR and logical product of said SS and SR signals.

21. A method for transmitting compressed information in a voice multiplexing device, a multi-media multiplexing device, and a switching system that transmits signaling information of a voice channel as a signal-send (SS) signal and a signal-receive (SR) signal, wherein:
said method for transmitting compressed information transmits a transmission frame with a flag indicating whether or not said SS signal is to be transmitted for a certain transmission cycle unit, and to compress the transmission of signaling information where said SS signal remains unchanged.

22. A method for transmitting compressed information in a multi-media multiplexing device or a switching system for transmitting a level signal for accommodated data channels to confirm the mode between a correspondent as a remote signaling (RS) signal, wherein:
said method for transmitting compressed information is used to transmit a transmission frame with a flag indicating whether or not said RS signal is to be transmitted for a certain transmission cycle unit, and to compress the transmission of remote signaling information where said RS signal does not change.

23. A high performance digitally multiplexed transmission system for use in a multiplexing device for

transmitting multiplexed signals on a digital line, said high performance digitally multiplexed transmission system comprising:

a voice coding means for separating voice input information into a core information part for ensuring minimum acceptable sound quality and, in a supplementary information part, for sending supplementary information to be sent with said core information part for getting desirable sound quality, parts of said supplementary information part discardable in sequence according to the priorities, said voice coding means for coding said core information part and said supplementary information part; and

a multiplexing means for multiplexing said core information part for a fixed length digital slot repeated at a predetermined cycle for a channel where said silence section detecting means detects no silence sections, for multiplexing said supplementary information part of each voice channel in the order of higher priority where no silence sections are detected for the remaining part of said fixed length digital slot, and for discarding a supplementary information part which cannot be multiplexed due to insufficient band.

24. A high performance digitally multiplexed transmission system for use in a switching system for transmitting multiplexed signals on a digital line, said high performance digitally multiplexed transmission system comprising:

a voice coding means for separating voice input information into a core information part for ensuring minimum acceptable sound quality and, in a supplementary information part, for sending supplementary information to be sent with said core information part for getting desirable sound quality, parts of said supplementary information part discardable in sequence according to the priorities, said voice coding means for coding said core information part and said supplementary information part;

a detecting means for detecting a silence section in said voice input information; and

a multiplexing means for multiplexing, among a plurality of voice channels to be multiplexed, only information necessary for synchronization with a correspondent's coder for a voice channel where said silence section detecting means detects a silence section, for multiplexing said core information part for a fixed length digital slot repeated at a predetermined cycle for a channel where said silence section detecting means detects no silence sections, for multiplexing said supplementary information part of each voice channel in the order of higher priority where no silence sections are detected for the remaining part of said fixed length digital slot, and for discarding a supplementary information part which cannot be multiplexed due to insufficient band.

25. A high performance digitally multiplexed transmission system for use in a multiplexing device for transmitting multiplexed signals on a digital line, said high performance digitally multiplexed transmission system comprising:

an image coding means for separating image input information into a core information part for ensuring minimum acceptable image quality and, in a supplementary information part, for sending supplementary information to be sent with said core information part for getting desirable image quality, parts of said supplementary information part discardable in sequence according to the priorities, said image coding means for coding said core information part and said supplementary information part;

a request band determining means for determining a transmission band of said image input information for determining the band necessary for transmitting image information;

a multiplexing means for multiplexing, among a plurality of image channels to be multiplexed, only information necessary for synchronization with a correspondent's coder for an image channel where said request band determining means determines it is necessary to transmit images, for multiplexing said core information part for a fixed length digital slot repeated at a predetermined cycle for a channel where said request band determining means determines it is necessary to transmit images, for multiplexing said supplementary information part of each image channel in the order of higher priority where it is determined necessary to transmit images for the remaining part of said fixed length digital slot, and for discarding a supplementary information part which cannot be multiplexed due to insufficient band.

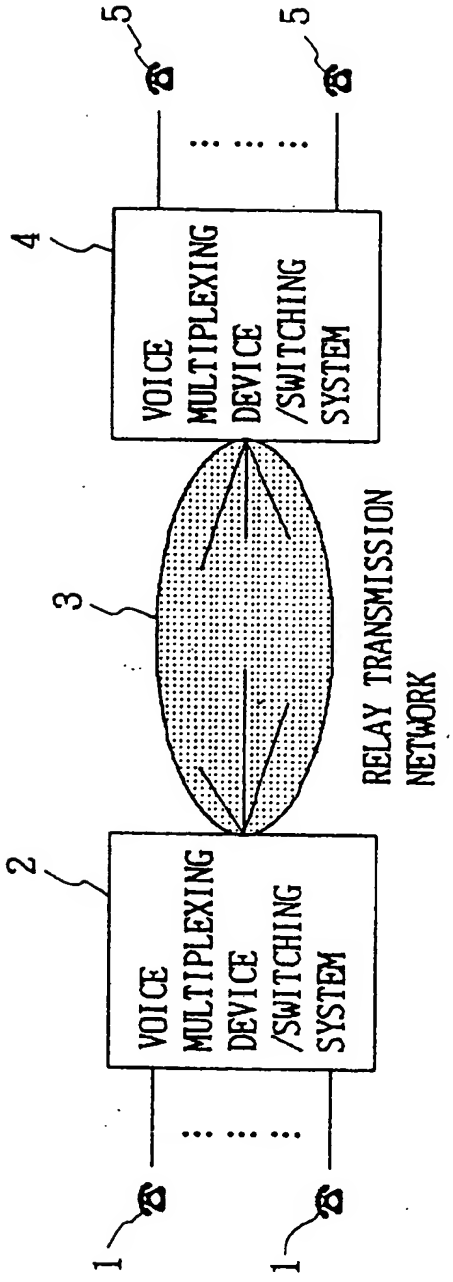

5
5
4
VOICE MULTIPLEXING DEVICE /SWITCHING SYSTEM
3
RELAY TRANSMISSION NETWORK
2
VOICE MULTIPLEXING DEVICE /SWITCHING SYSTEM
1
1

Fig. 1

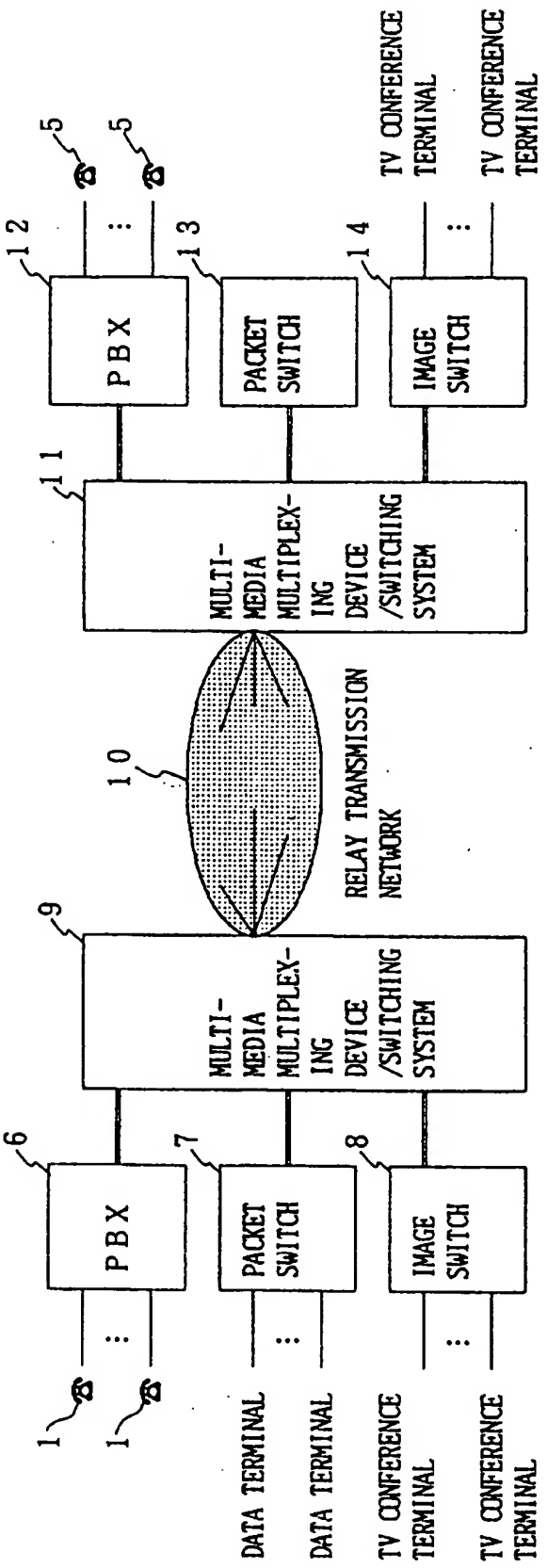

1
1
DATA TERMINAL
DATA TERMINAL
TV CONFERENCE TERMINAL
TV CONFERENCE TERMINAL
6
PBX
7
PACKET SWITCH
8
IMAGE SWITCH
9
MULTI-MEDIA MULTIPLEX-ING DEVICE /SWITCHING SYSTEM
10
RELAY TRANSMISSION NETWORK
11
MULTI-MEDIA MULTIPLEX-ING DEVICE /SWITCHING SYSTEM
12
PBX
13
PACKET SWITCH
14
IMAGE SWITCH
5
5
TV CONFERENCE TERMINAL
TV CONFERENCE TERMINAL

Fig. 2

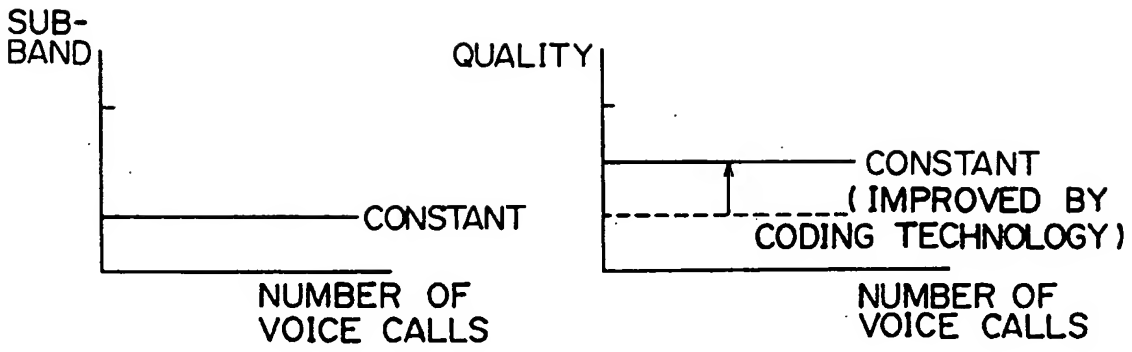

SUB-
BAND
CONSTANT
NUMBER OF
VOICE CALLS
QUALITY
CONSTANT
(IMPROVED BY
CODING TECHNOLOGY)
NUMBER OF
VOICE CALLS

Fig. 3A

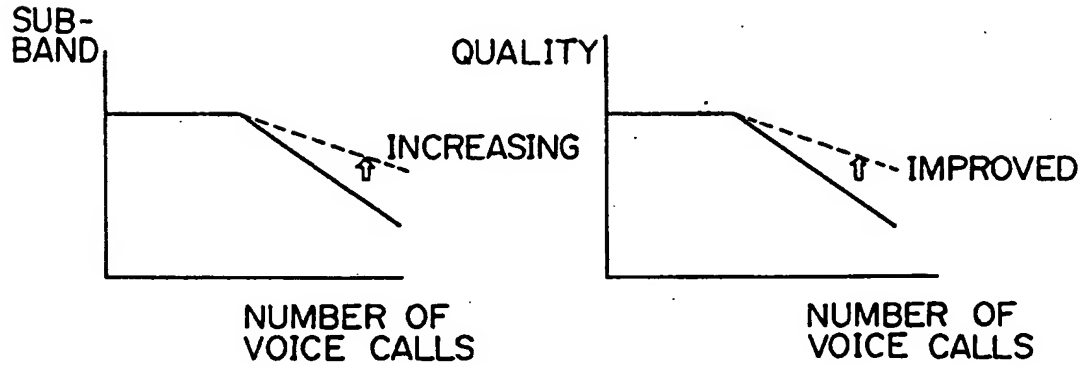

SUB-
BAND
INCREASING
NUMBER OF
VOICE CALLS
QUALITY
IMPROVED
NUMBER OF
VOICE CALLS

Fig. 3B

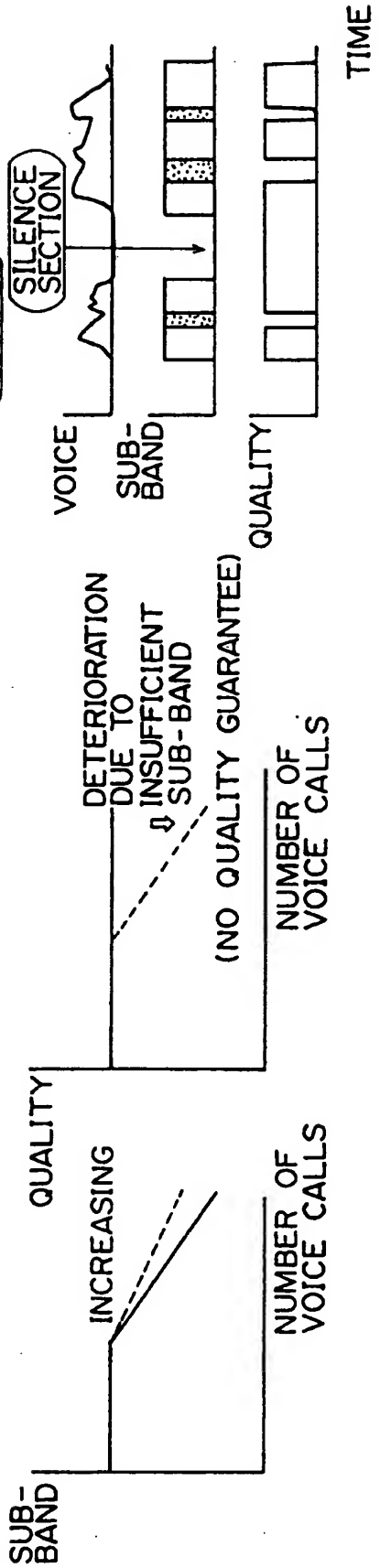

TIME
DISCARD
SILENCE SECTION
VOICE
SUB-BAND
QUALITY
TIME
DETERIORATION DUE TO INSUFFICIENT SUB-BAND
(NO QUALITY GUARANTEE)
QUALITY
NUMBER OF VOICE CALLS
INCREASING
QUALITY
NUMBER OF VOICE CALLS
SUB-BAND

Fig. 4

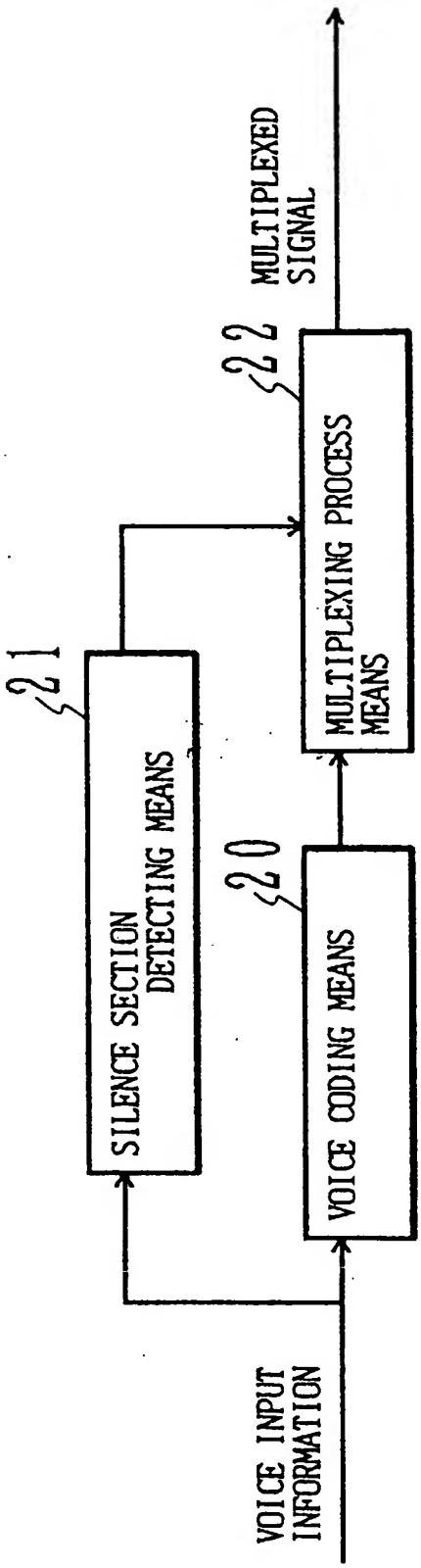

VOICE INPUT INFORMATION
20
VOICE CODING MEANS
21
SILENCE SECTION DETECTING MEANS
22
MULTIPLEXING PROCESS MEANS
MULTIPLEXED SIGNAL

Fig. 5A

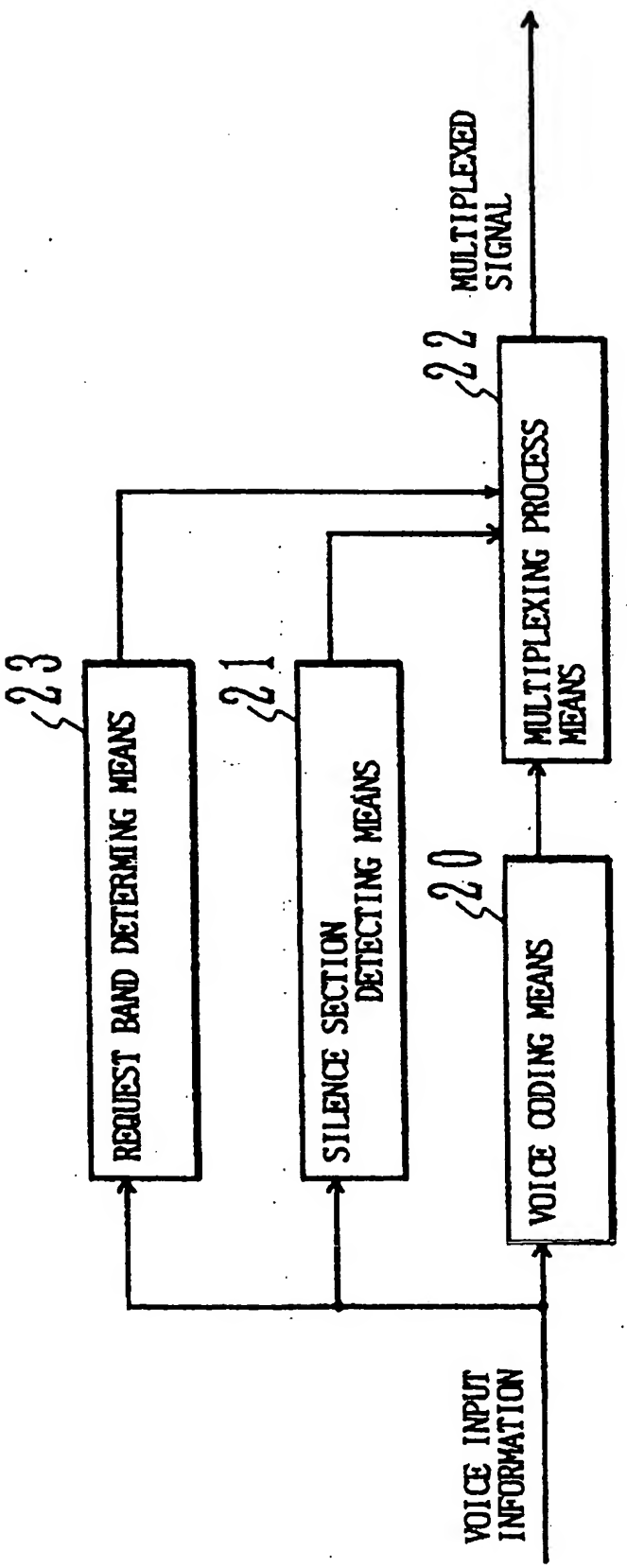

VOICE INPUT INFORMATION
20
VOICE CODING MEANS
21
SILENCE SECTION DETECTING MEANS
23
REQUEST BAND DETERMING MEANS
22
MULTIPLEXING PROCESS MEANS
MULTIPLEXED SIGNAL

Fig. 5B

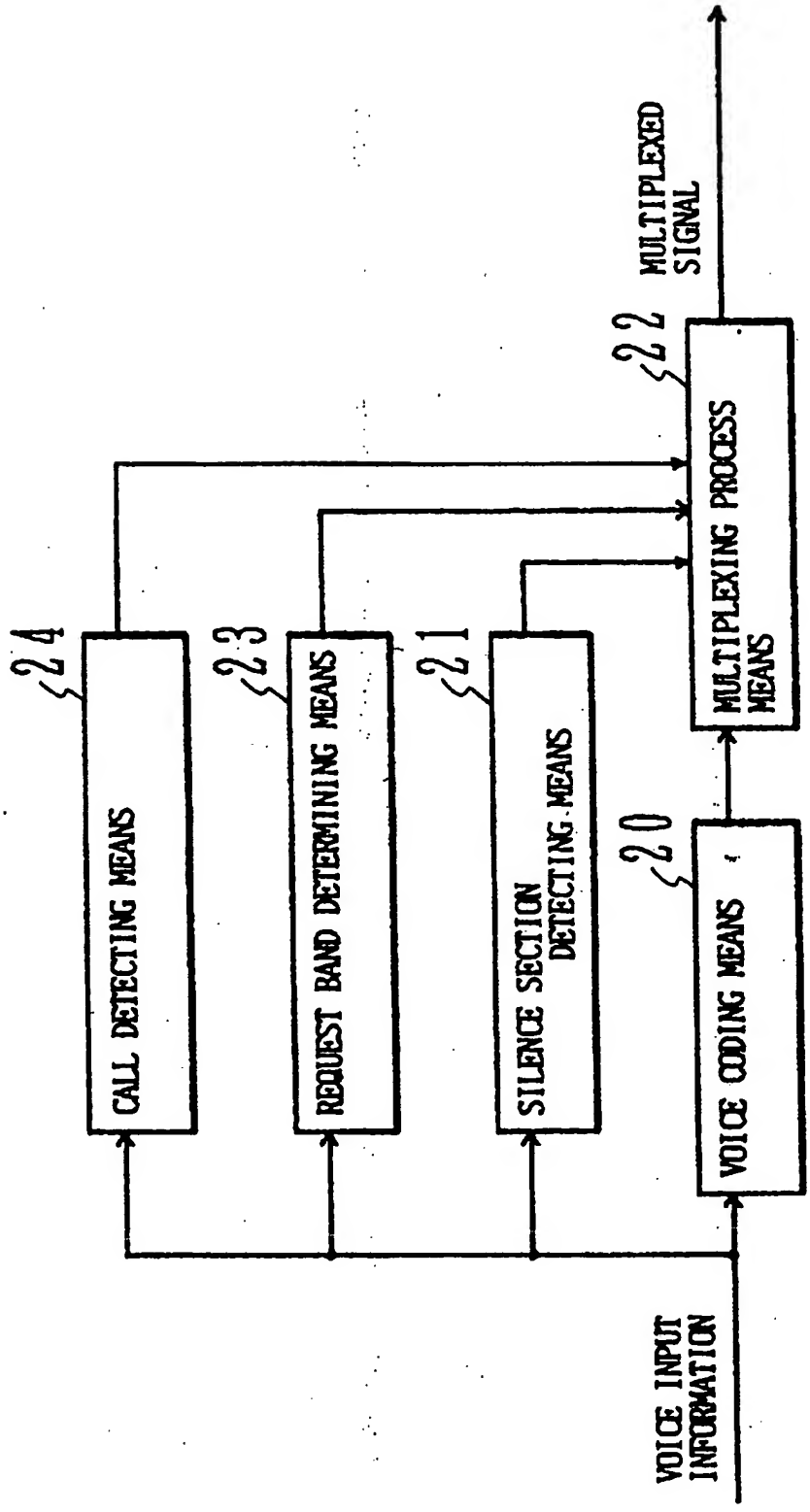

24
CALL DETECTING MEANS
23
REQUEST BAND DETERMINING MEANS
21
SILENCE SECTION DETECTING MEANS
20
VOICE CODING MEANS
VOICE INPUT INFORMATION
MULTIPLEXING PROCESS MEANS
22
MULTIPLEXED SIGNAL

Fig. 5C

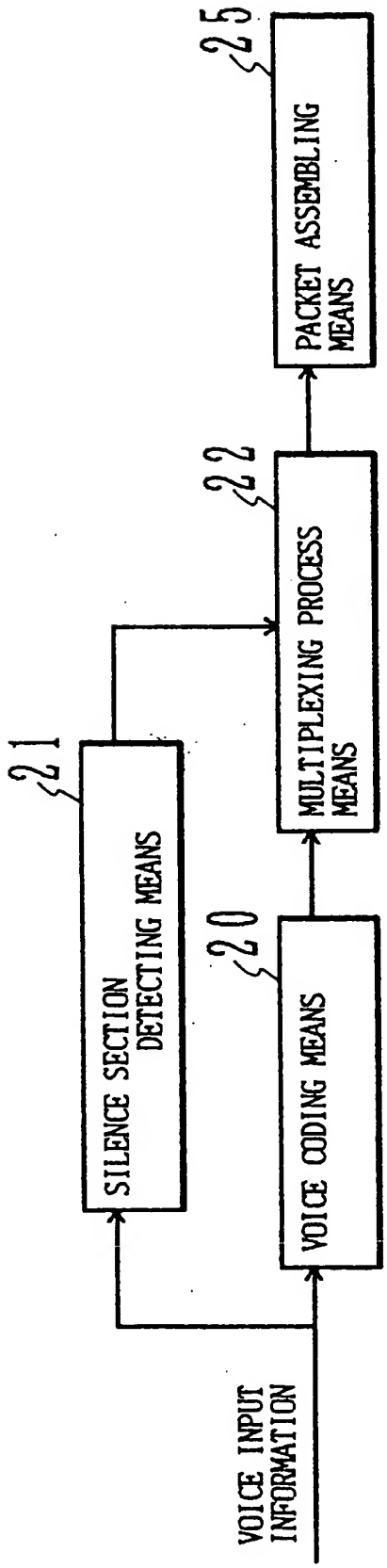

VOICE INPUT INFORMATION
20
VOICE CODING MEANS
21
SILENCE SECTION DETECTING MEANS
22
MULTIPLEXING PROCESS MEANS
25
PACKET ASSEMBLING MEANS

Fig. 5D

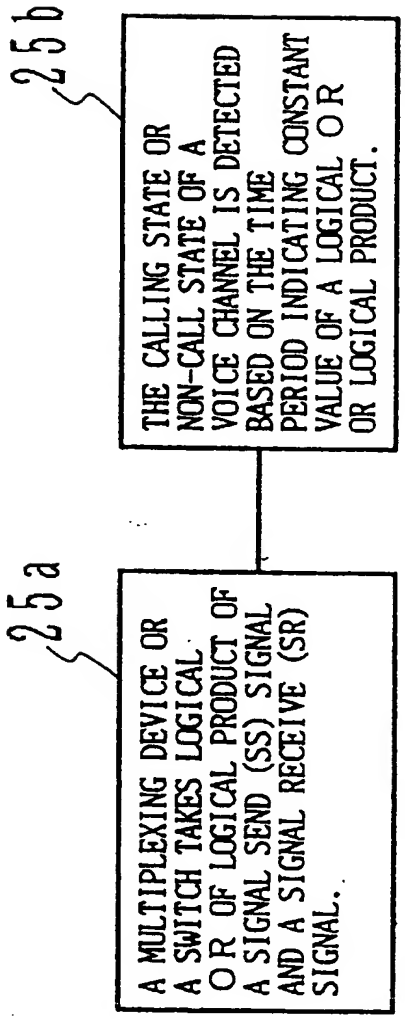

25a
A MULTIPLEXING DEVICE OR A SWITCH TAKES LOGICAL OR OF LOGICAL PRODUCT OF A SIGNAL SEND (SS) SIGNAL AND A SIGNAL RECEIVE (SR) SIGNAL.
25b
THE CALLING STATE OR NON-CALL STATE OF A VOICE CHANNEL IS DETECTED BASED ON THE TIME PERIOD INDICATING CONSTANT VALUE OF A LOGICAL OR OR LOGICAL PRODUCT.

Fig. 5E

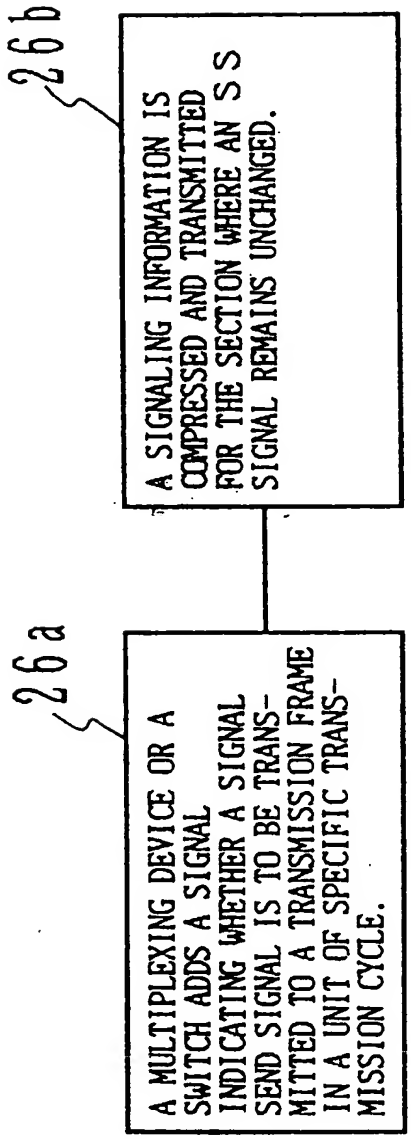

26b
A SIGNALING INFORMATION IS COMPRESSED AND TRANSMITTED FOR THE SECTION WHERE AN S S SIGNAL REMAINS UNCHANGED.
26a
A MULTIPLEXING DEVICE OR A SWITCH ADDS A SIGNAL INDICATING WHETHER A SIGNAL SEND SIGNAL IS TO BE TRANS-MITTED TO A TRANSMISSION FRAME IN A UNIT OF SPECIFIC TRANS-MISSION CYCLE.

Fig. 5F

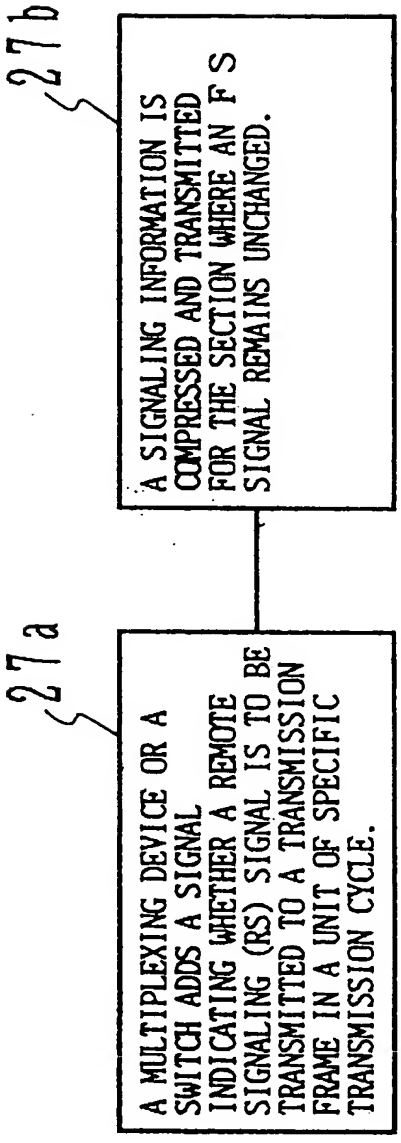

27a
A MULTIPLEXING DEVICE OR A SWITCH ADDS A SIGNAL INDICATING WHETHER A REMOTE SIGNALING (RS) SIGNAL IS TO BE TRANSMITTED TO A TRANSMISSION FRAME IN A UNIT OF SPECIFIC TRANSMISSION CYCLE.
27b
A SIGNALING INFORMATION IS COMPRESSED AND TRANSMITTED FOR THE SECTION WHERE AN F S SIGNAL REMAINS UNCHANGED.

Fig. 5G

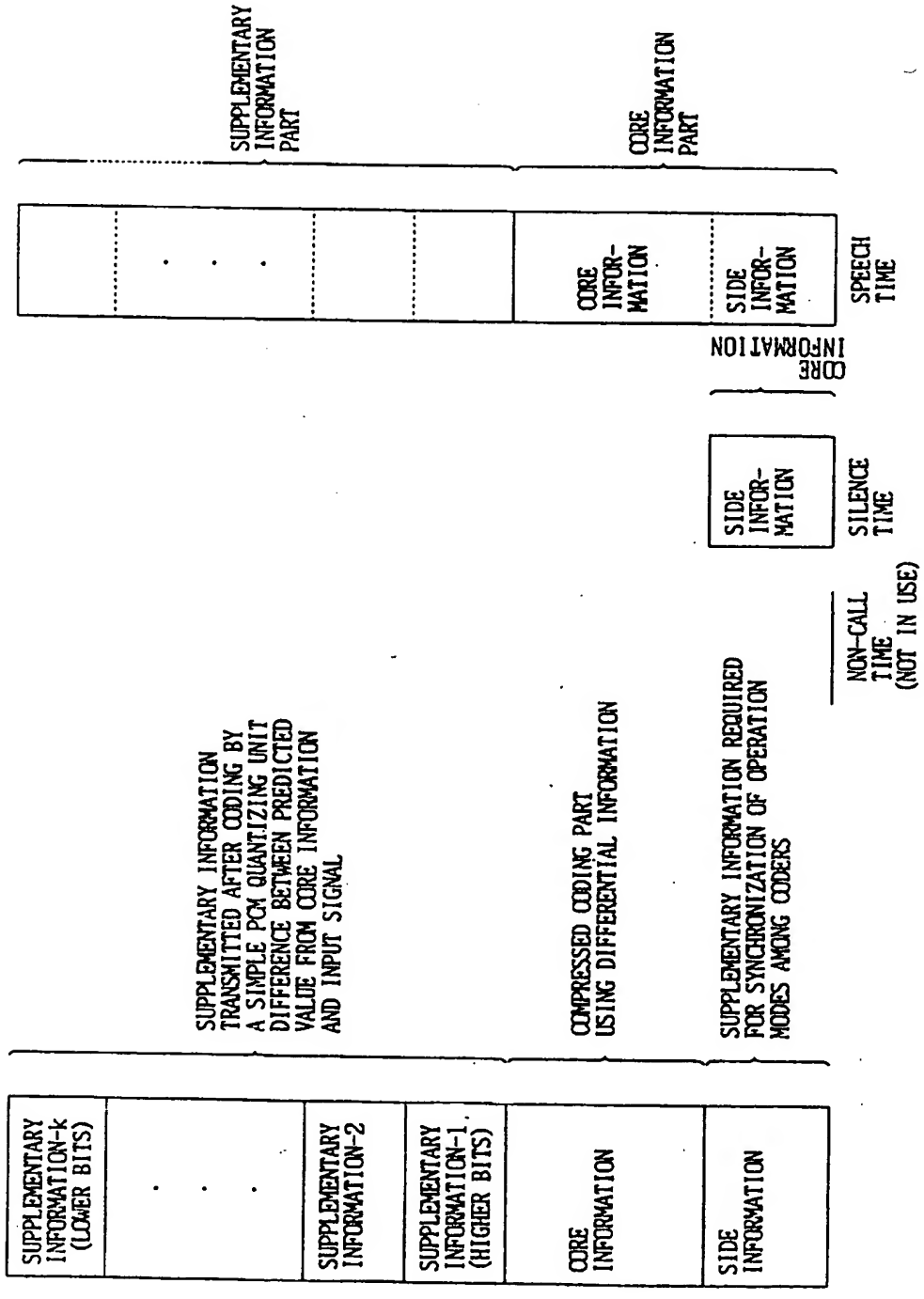

SUPPLEMENTARY INFORMATION-k (LOWER BITS)
SUPPLEMENTARY INFORMATION-2
SUPPLEMENTARY INFORMATION-1 (HIGHER BITS)
CORE INFORMATION
SIDE INFORMATION
SUPPLEMENTARY INFORMATION TRANSMITTED AFTER CODING BY A SIMPLE PCM QUANTIZING UNIT DIFFERENCE BETWEEN PREDICTED VALUE FROM CORE INFORMATION AND INPUT SIGNAL
COMPRESSED CODING PART USING DIFFERENTIAL INFORMATION
SUPPLEMENTARY INFORMATION REQUIRED FOR SYNCHRONIZATION OF OPERATION MODES AMONG CODERS
NON-CALL TIME (NOT IN USE)
SIDE INFOR-MATION
SILENCE TIME
CORE INFORMATION
CORE INFOR-MATION
SIDE INFOR-MATION
SPEECH TIME
CORE INFORMATION PART
SUPPLEMENTARY INFORMATION PART

Fig. 6A

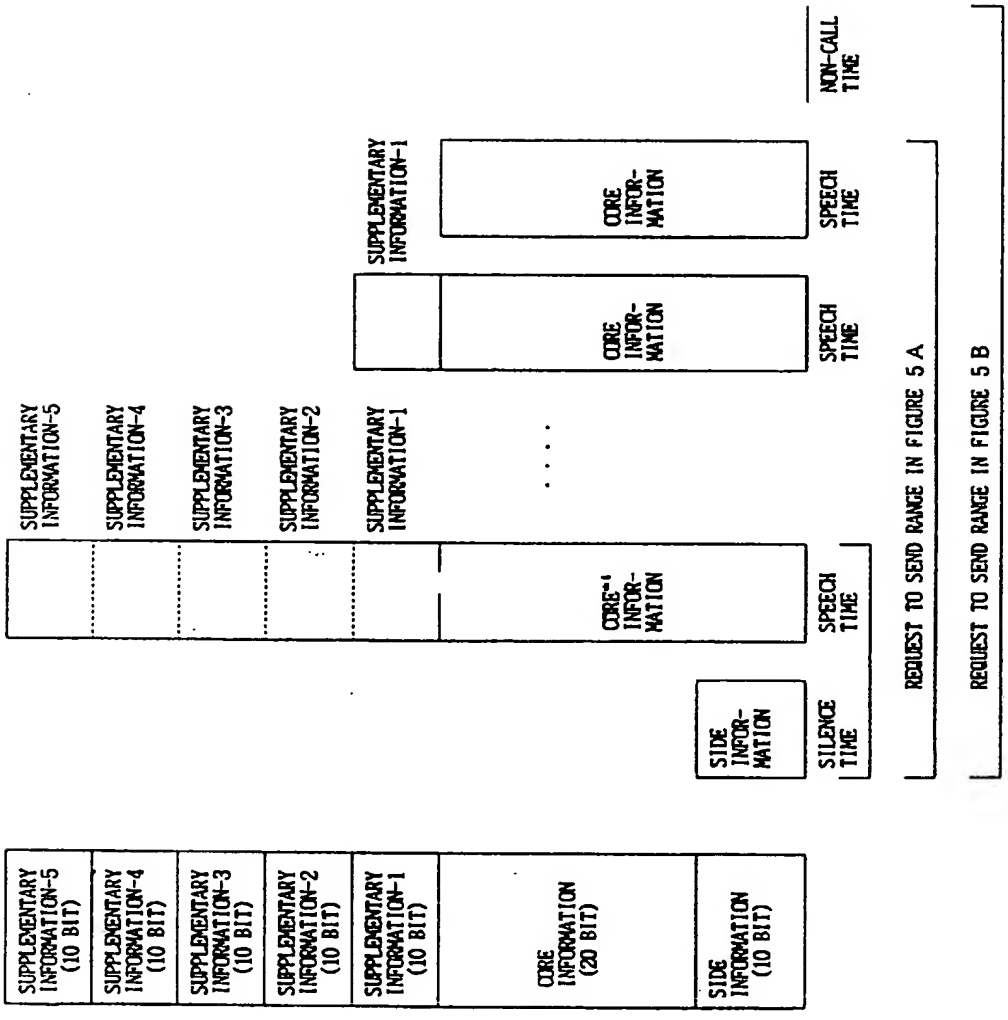

SUPPLEMENTARY INFORMATION-5 (10 BIT)
SUPPLEMENTARY INFORMATION-4 (10 BIT)
SUPPLEMENTARY INFORMATION-3 (10 BIT)
SUPPLEMENTARY INFORMATION-2 (10 BIT)
SUPPLEMENTARY INFORMATION-1 (10 BIT)
CORE INFORMATION (20 BIT)
SIDE INFORMATION (10 BIT)
SUPPLEMENTARY INFORMATION-5
SUPPLEMENTARY INFORMATION-4
SUPPLEMENTARY INFORMATION-3
SUPPLEMENTARY INFORMATION-2
SUPPLEMENTARY INFORMATION-1
CORE-1 INFOR-MATION
SIDE INFOR-MATION
SILENCE TIME
SPEECH TIME
SUPPLEMENTARY INFORMATION-1
CORE INFOR-MATION
SPEECH TIME
SUPPLEMENTARY INFORMATION-1
CORE INFOR-MATION
SPEECH TIME
NON-CALL TIME
REQUEST TO SEND RANGE IN FIGURE 5 A
REQUEST TO SEND RANGE IN FIGURE 5 B
REQUEST TO SEND RANGE IN FIGURE 5 C

Fig. 6B

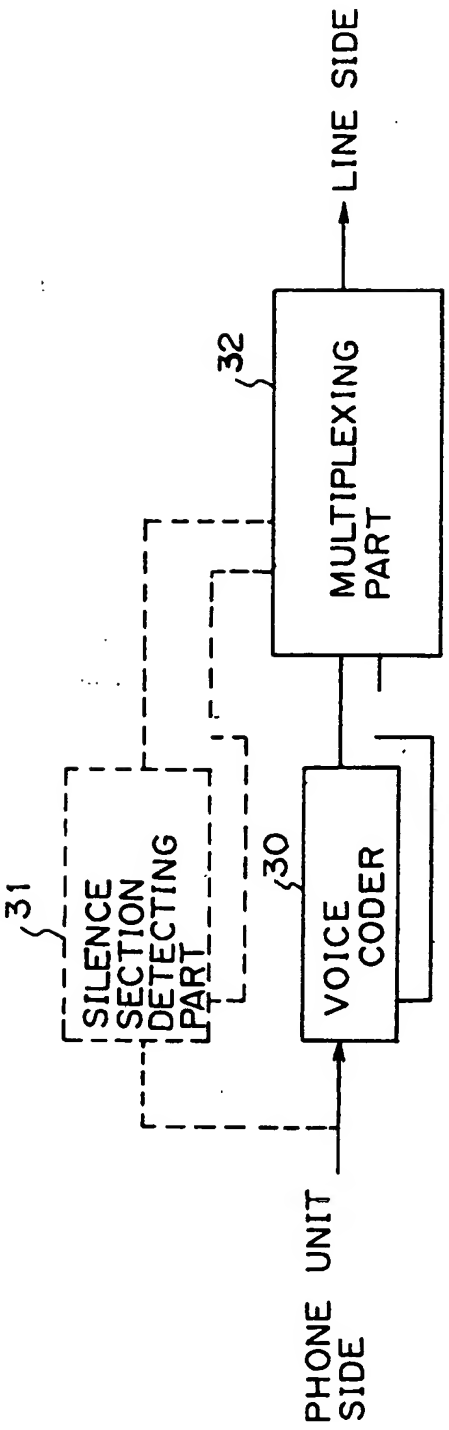

LINE SIDE
32
MULTIPLEXING PART
31
SILENCE SECTION DETECTING PART
30
VOICE CODER
PHONE UNIT SIDE

Fig. 7A

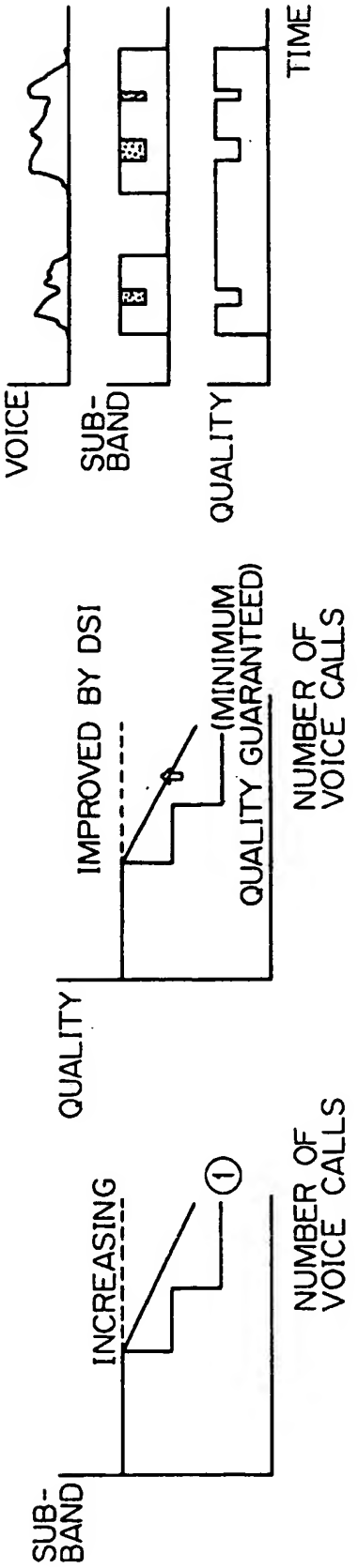

VOICE
SUB-BAND
QUALITY
TIME
SUB-BAND
INCREASING
①
NUMBER OF VOICE CALLS
QUALITY
IMPROVED BY DSI
(MINIMUM QUALITY GUARANTEED)
NUMBER OF VOICE CALLS

Fig. 7B

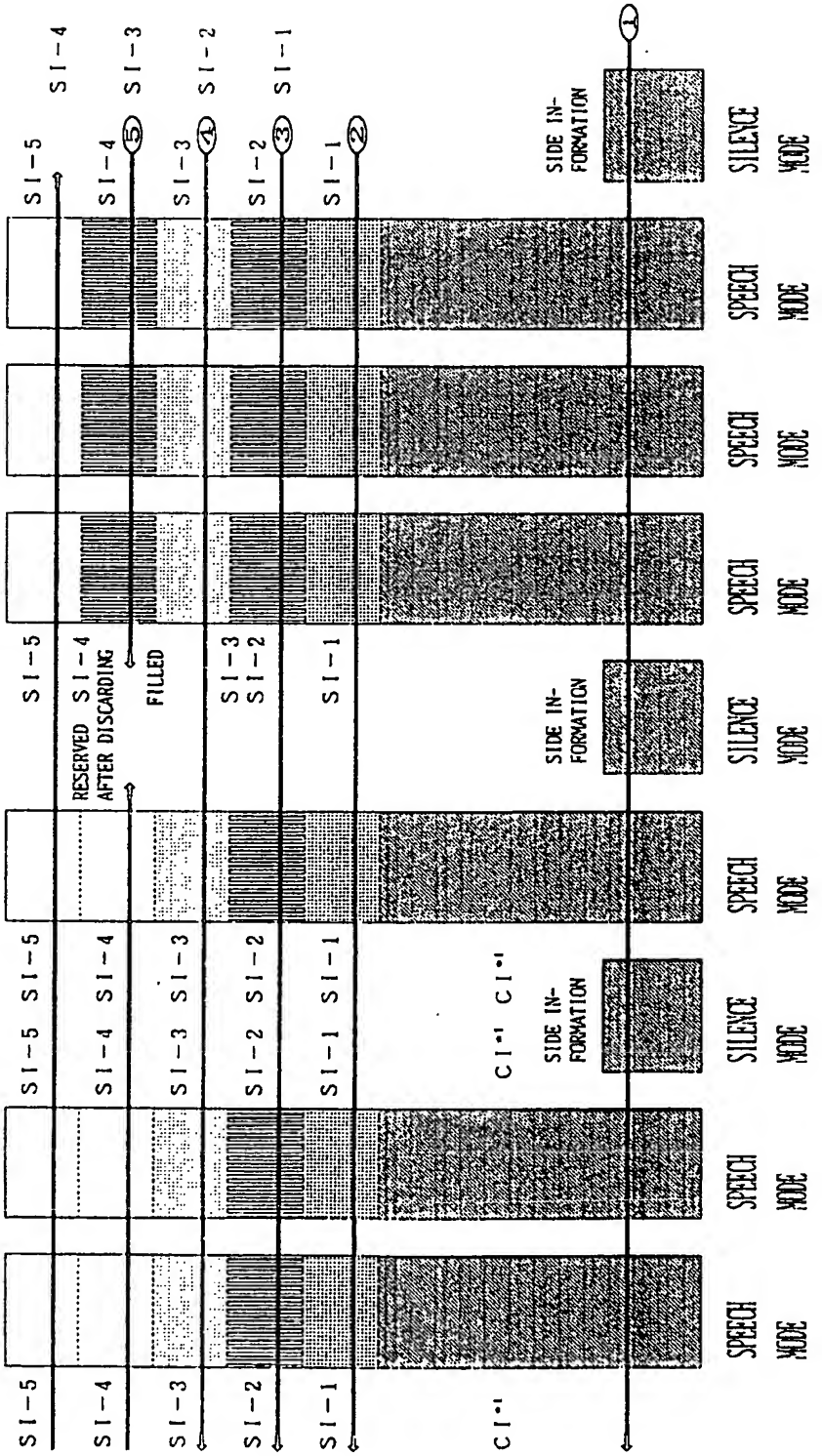

CH #0
CH #1
CH #2
CH #3
CH #4
CH #5
CH #6
CH #7
SI-5
SI-4
SI-3
SI-2
SI-1
CI
SPEECH MODE
SILENCE MODE
RESERVED SI-4 AFTER DISCARDING
FILLED
SIDE INFORMATION
SI: SUPPLEMENTARY INFORMATION
CI: CORE INFORMATION

Fig. 7C

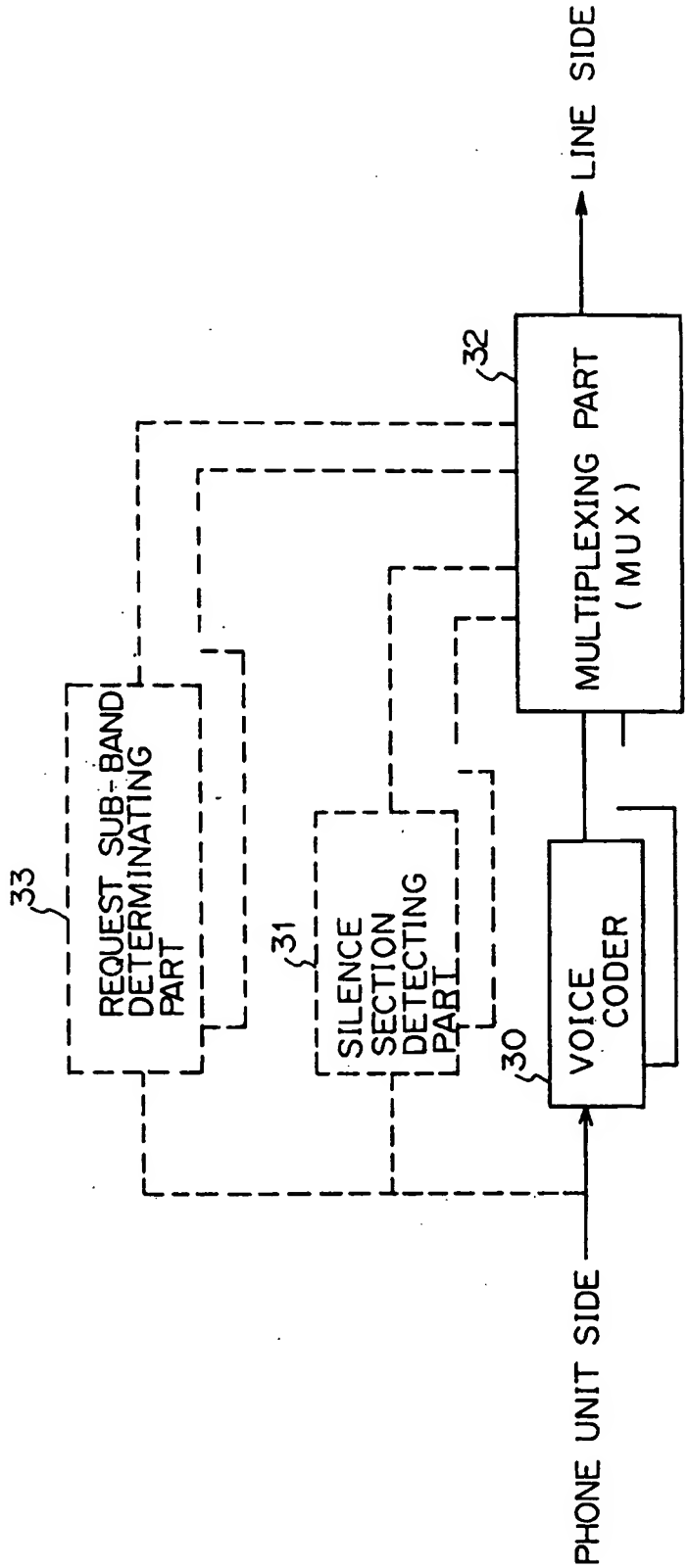

LINE SIDE
32
MULTIPLEXING PART
(MUX)
33
REQUEST SUB-BAND
DETERMINATING
PART
31
SILENCE
SECTION
DETECTING
PART
30
VOICE
CODER
PHONE UNIT SIDE

Fig. 8A

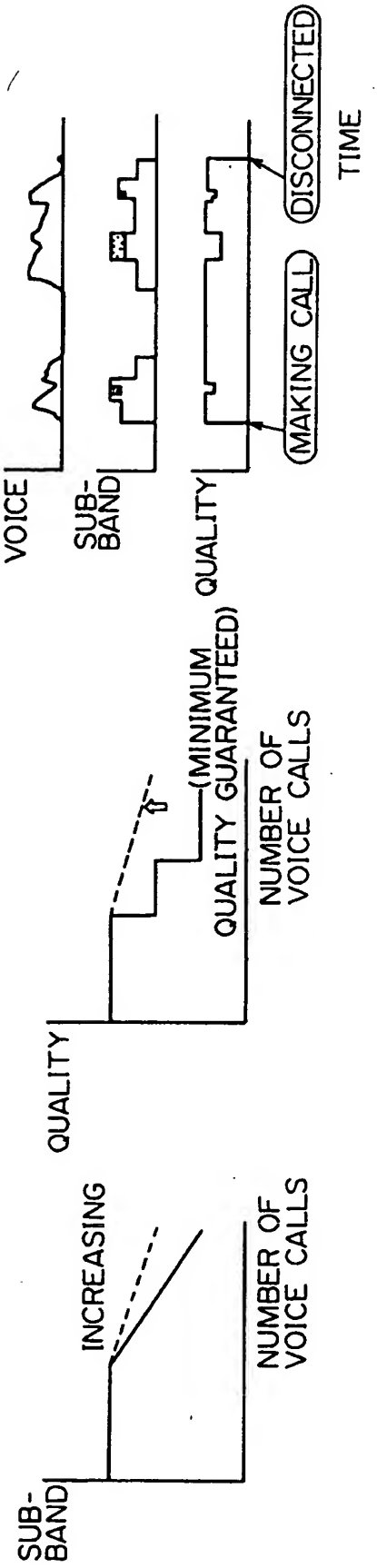

VOICE
SUB-BAND
QUALITY
MAKING CALL
DISCONNECTED
TIME
QUALITY
(MINIMUM QUALITY GUARANTEED)
NUMBER OF VOICE CALLS
SUB-BAND
INCREASING
NUMBER OF VOICE CALLS

Fig. 8B

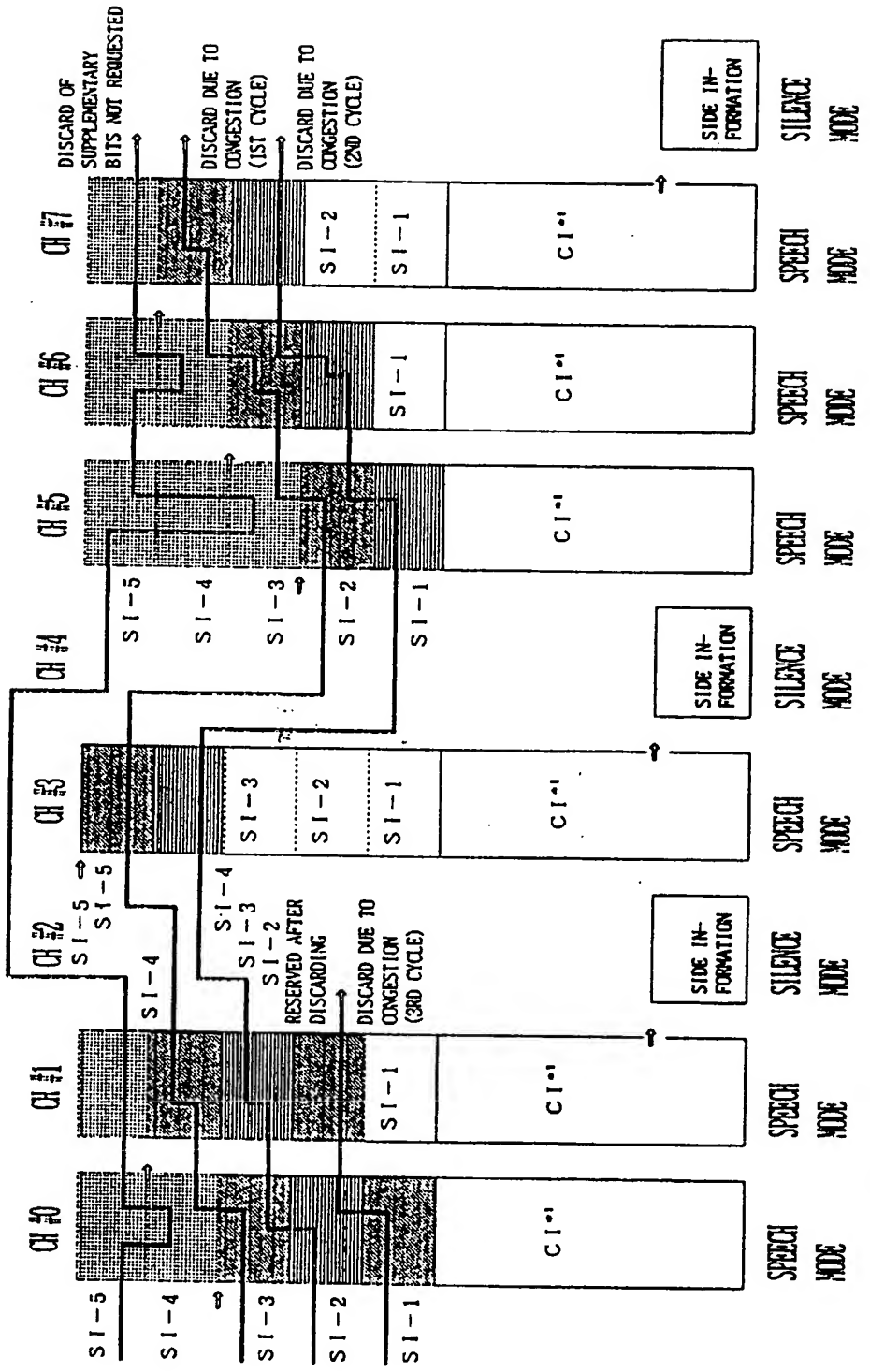

CH #0
CH #1
CH #2
CH #3
CH #4
CH #5
CH #6
CH #7
SI-5
SI-4
SI-3
SI-2
SI-1
CI
SPEECH MODE
SILENCE MODE
SIDE INFORMATION
DISCARD OF SUPPLEMENTARY BITS NOT REQUESTED
DISCARD DUE TO CONGESTION (1ST CYCLE)
DISCARD DUE TO CONGESTION (2ND CYCLE)
DISCARD DUE TO CONGESTION (3RD CYCLE)
RESERVED AFTER DISCARDING
SI : SUPPLEMENTARY INFORMATION
CI : CORE INFORMATION

Fig. 8C

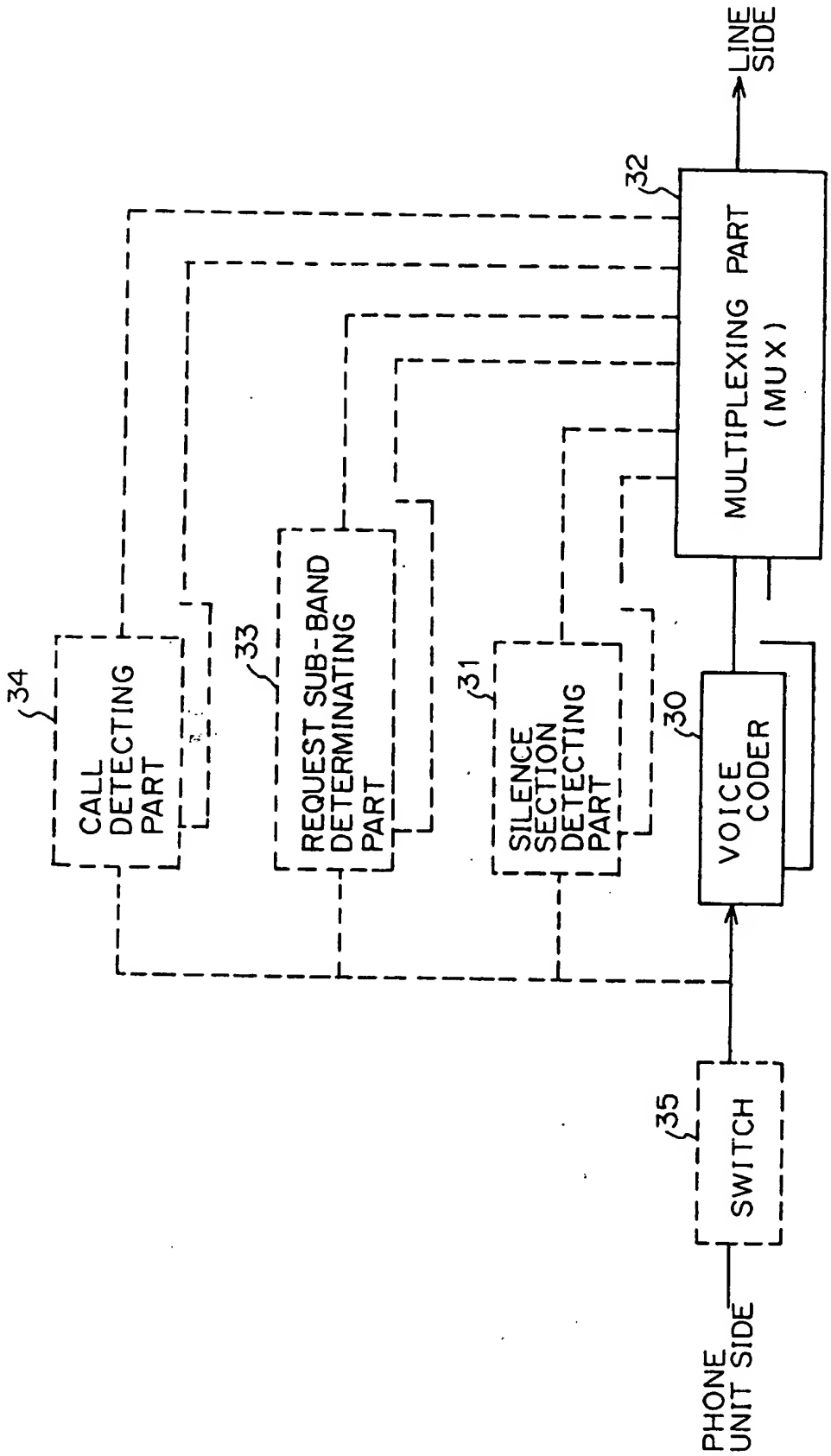

LINE SIDE
32
MULTIPLEXING PART (MUX)
34
CALL DETECTING PART
33
REQUEST SUB-BAND DETERMINATING PART
31
SILENCE SECTION DETECTING PART
30
VOICE CODER
35
SWITCH
PHONE UNIT SIDE

Fig. 9A

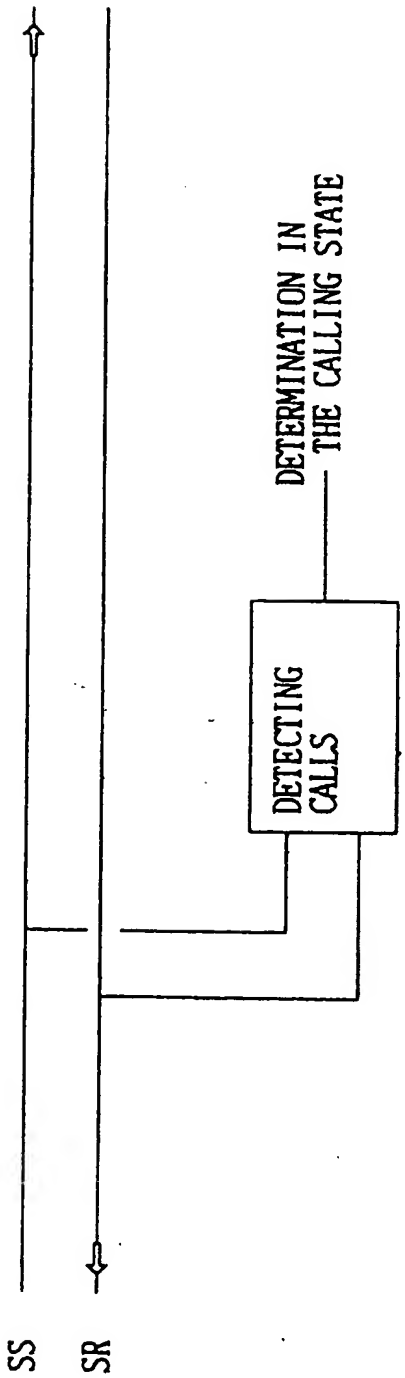
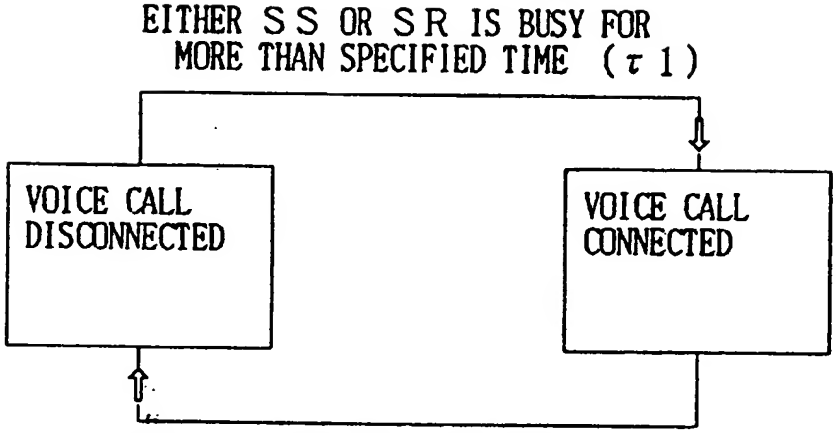


Fig. 9B



STANDARD OF SPECIFIED TIME: τ 1 ; 1 sec
τ 2 ; 1 sec

Fig. 9C

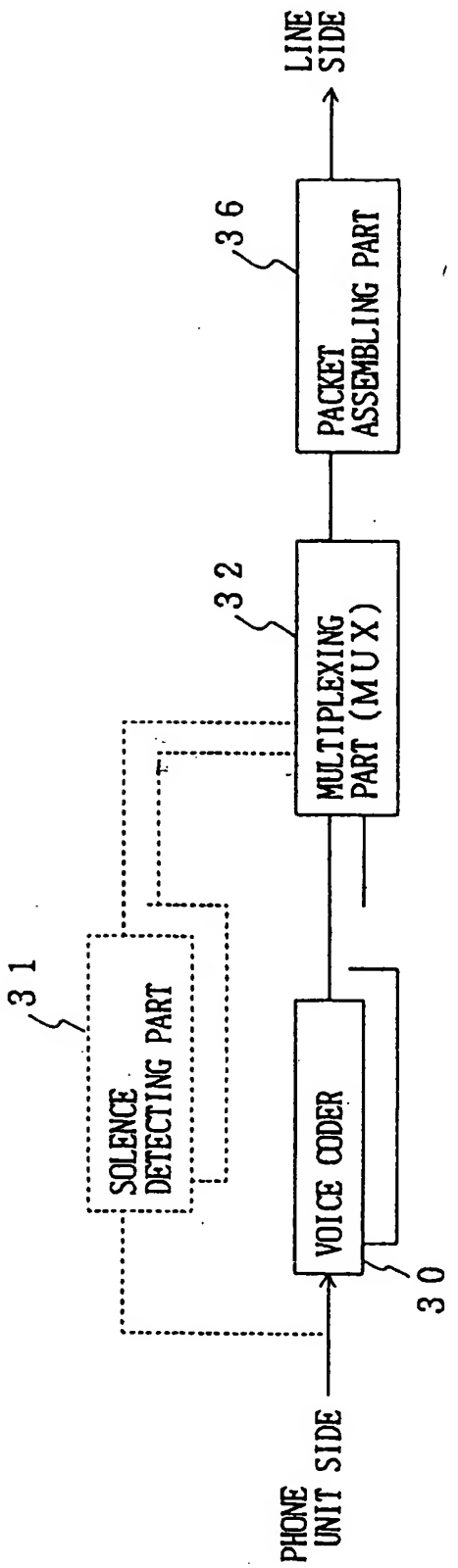

LINE SIDE
36
PACKET ASSEMBLING PART
32
MULTIPLEXING PART (MUX)
31
SOLENCE DETECTING PART
VOICE CODER
30
PHONE UNIT SIDE

Fig. 10

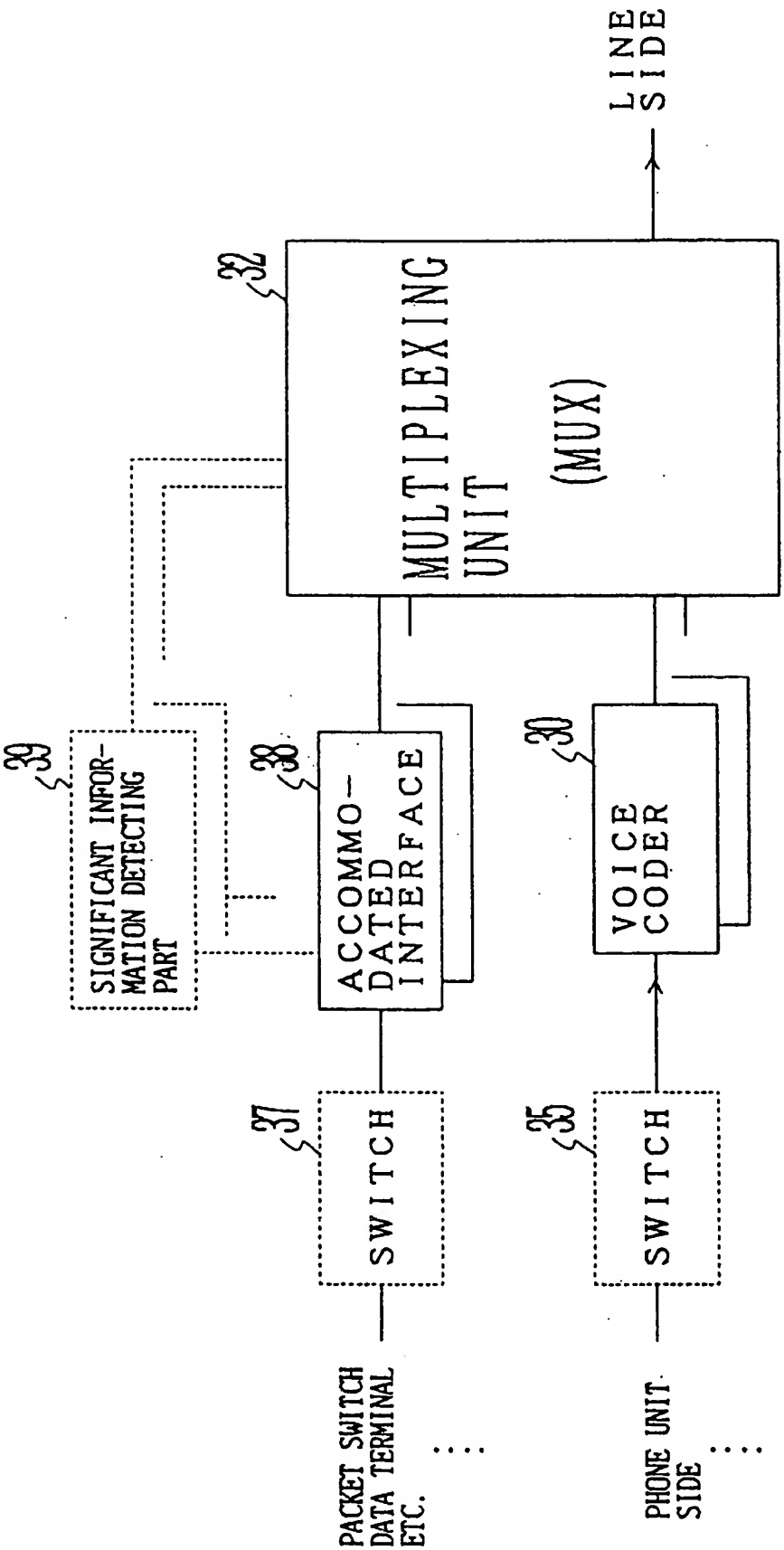

LINE SIDE
32
MULTIPLEXING UNIT
(MUX)
39
SIGNIFICANT INFOR-MATION DETECTING PART
38
ACCOMMO-DATED INTERFACE
30
VOICE CODER
37
SWITCH
35
SWITCH
PACKET SWITCH DATA TERMINAL ETC.
PHONE UNIT SIDE

Fig. 11

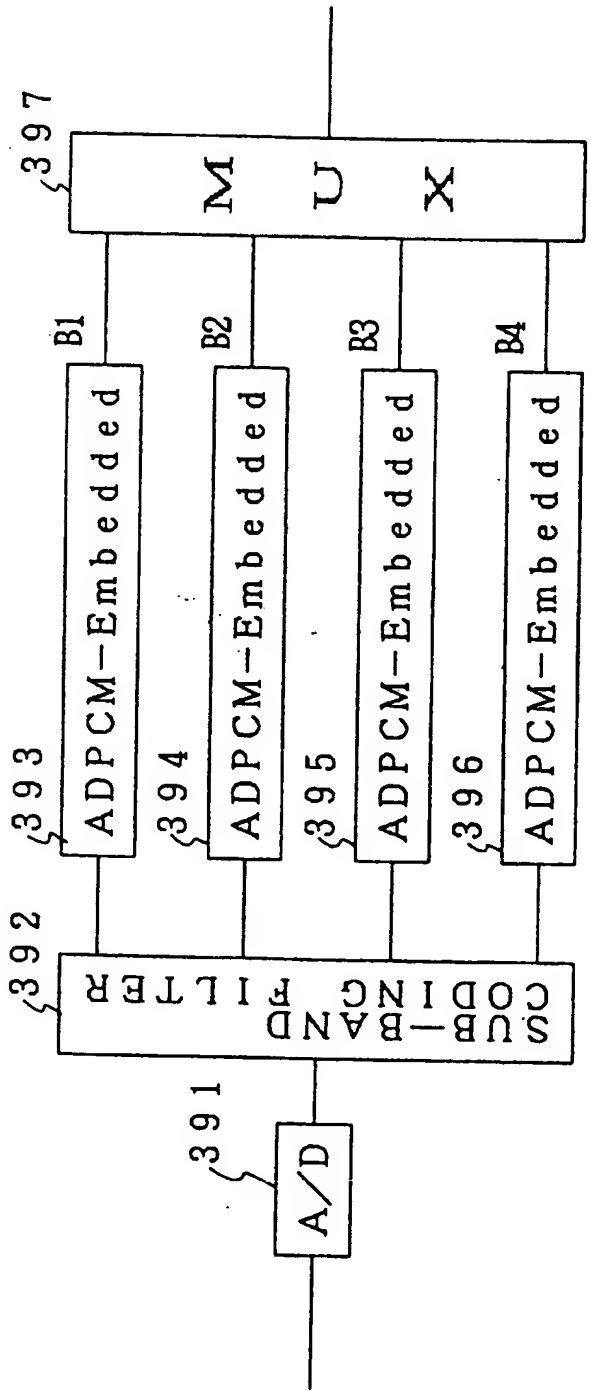

397
MUX
B1
B2
B3
B4
393
ADPCM-Embedded
394
ADPCM-Embedded
395
ADPCM-Embedded
396
ADPCM-Embedded
392
SUB-BAND CODING FILTER
391
A/D

Fig. 12A

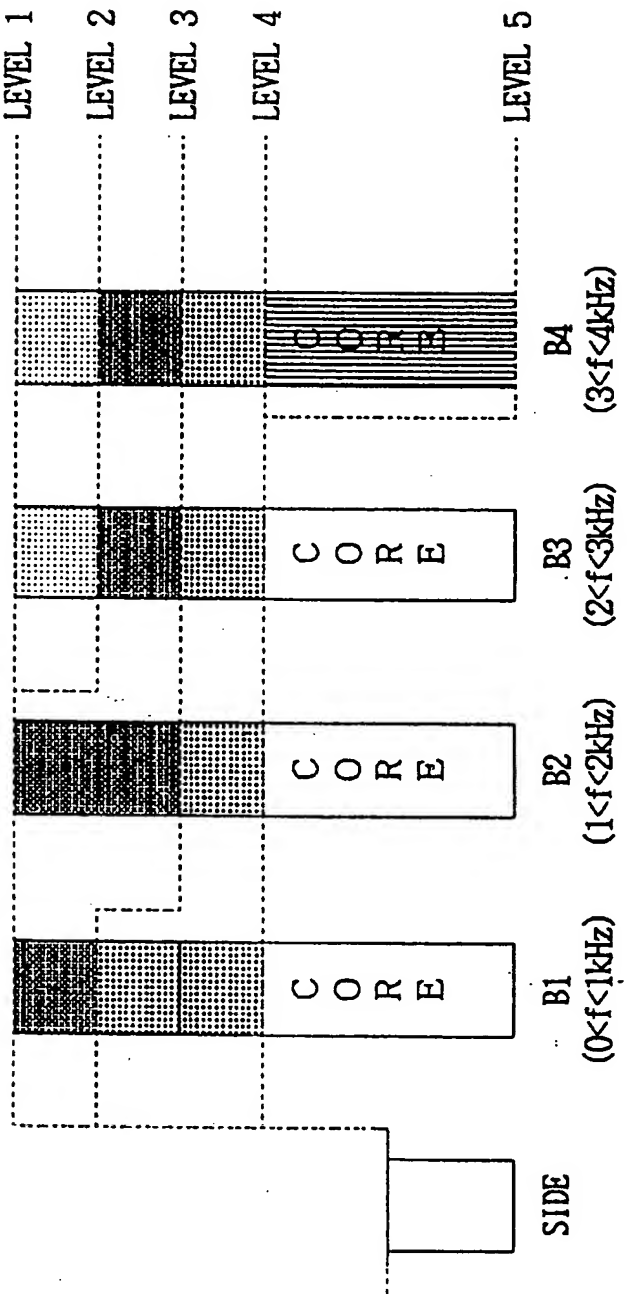

LEVEL 1
LEVEL 2
LEVEL 3
LEVEL 4
LEVEL 5
CORE
CORE
CORE
SIDE
B1 (0<f<1kHz)
B2 (1<f<2kHz)
B3 (2<f<3kHz)
B4 (3<f<4kHz)

Fig. 12B

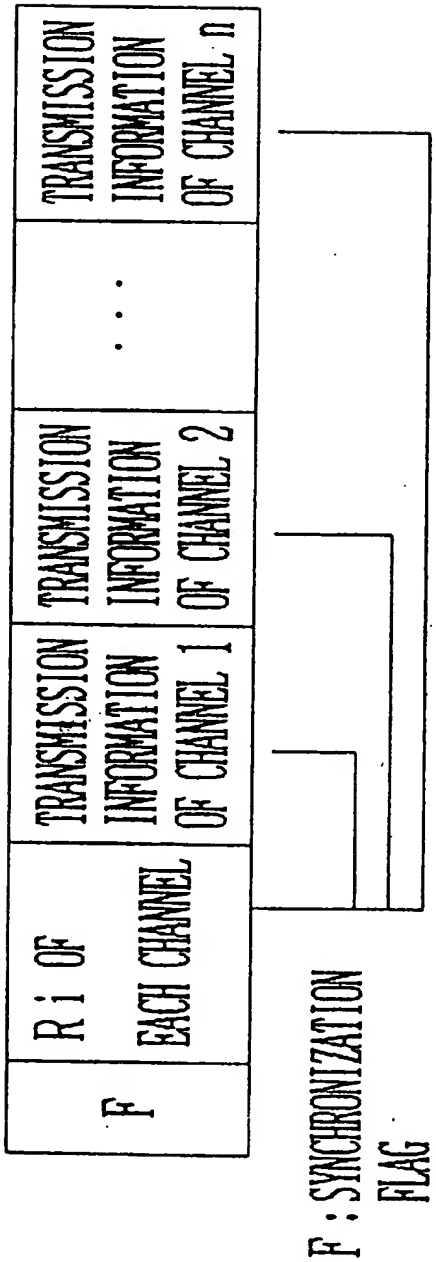

F
Ri OF EACH CHANNEL
TRANSMISSION INFORMATION OF CHANNEL 1
TRANSMISSION INFORMATION OF CHANNEL 2
. . .
TRANSMISSION INFORMATION OF CHANNEL n
F : SYNCHRONIZATION FLAG

Fig. 13

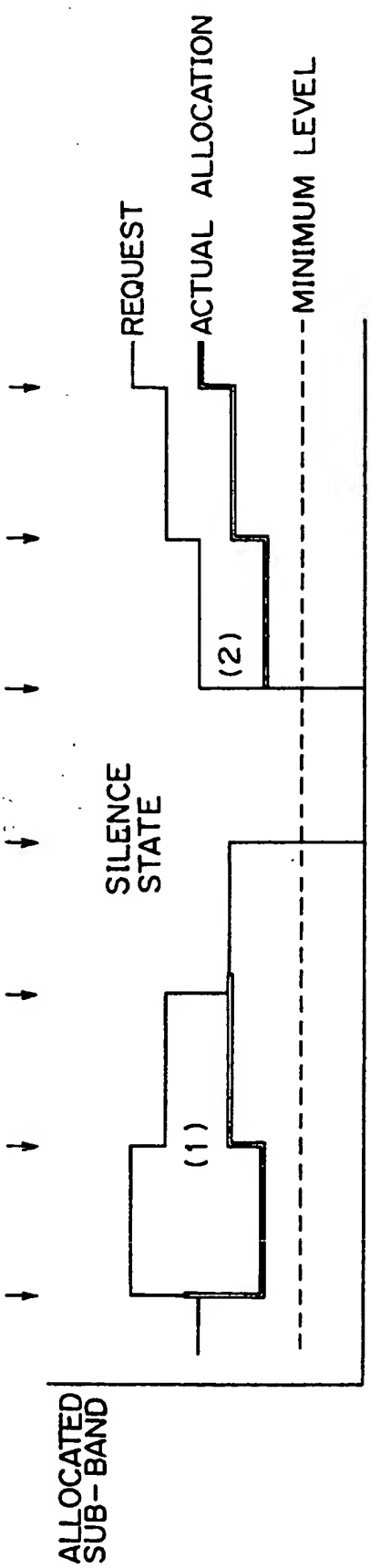

REQUEST
ACTUAL ALLOCATION
MINIMUM LEVEL
(2)
SILENCE STATE
(1)
ALLOCATED SUB-BAND

Fig. 14

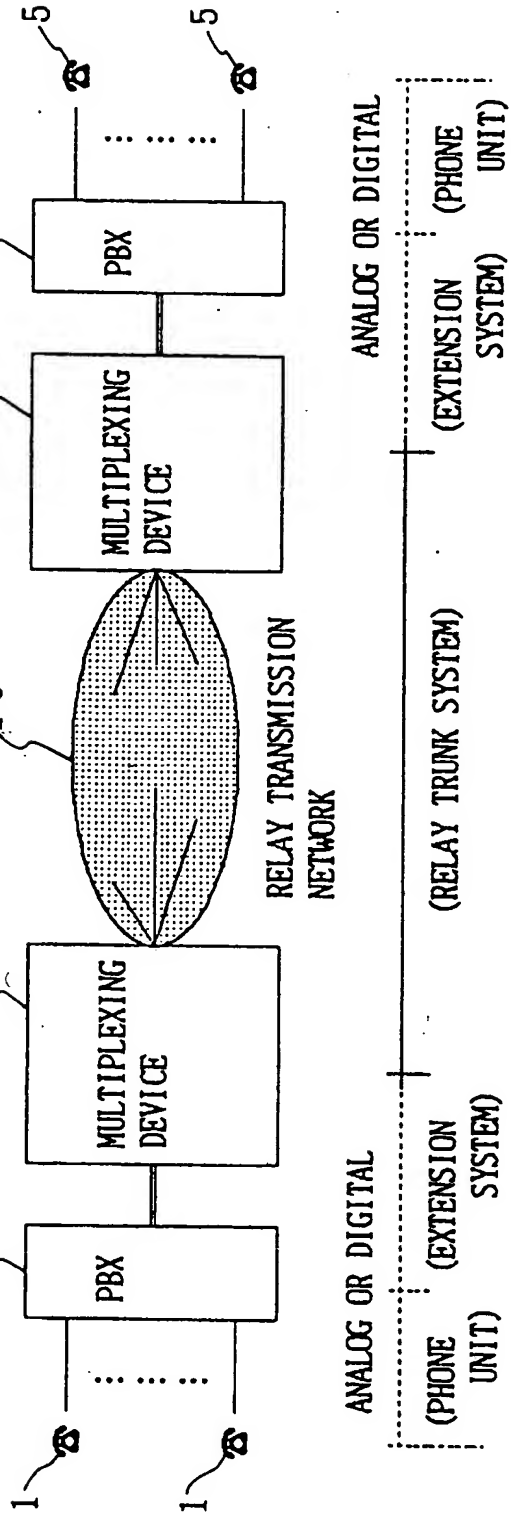

1
1
6
PBX
9
MULTIPLEXING DEVICE
10
RELAY TRANSMISSION NETWORK
11
MULTIPLEXING DEVICE
12
PBX
5
5
ANALOG OR DIGITAL (PHONE UNIT)
(EXTENSION SYSTEM)
(RELAY TRUNK SYSTEM)
(EXTENSION SYSTEM)
ANALOG OR DIGITAL (PHONE UNIT)

Fig. 15

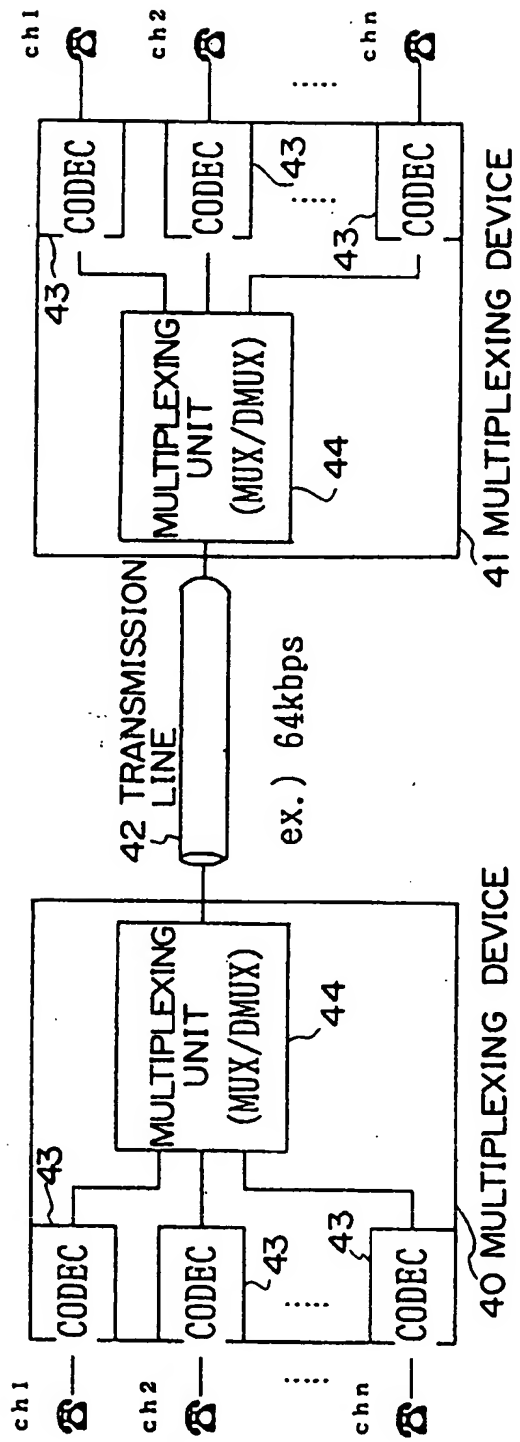

ch1
ch2
chn
CODEC
43
MULTIPLEXING UNIT (MUX/DMUX)
44
41 MULTIPLEXING DEVICE
42 TRANSMISSION LINE
ex.) 64kbps
40 MULTIPLEXING DEVICE

Fig. 16A

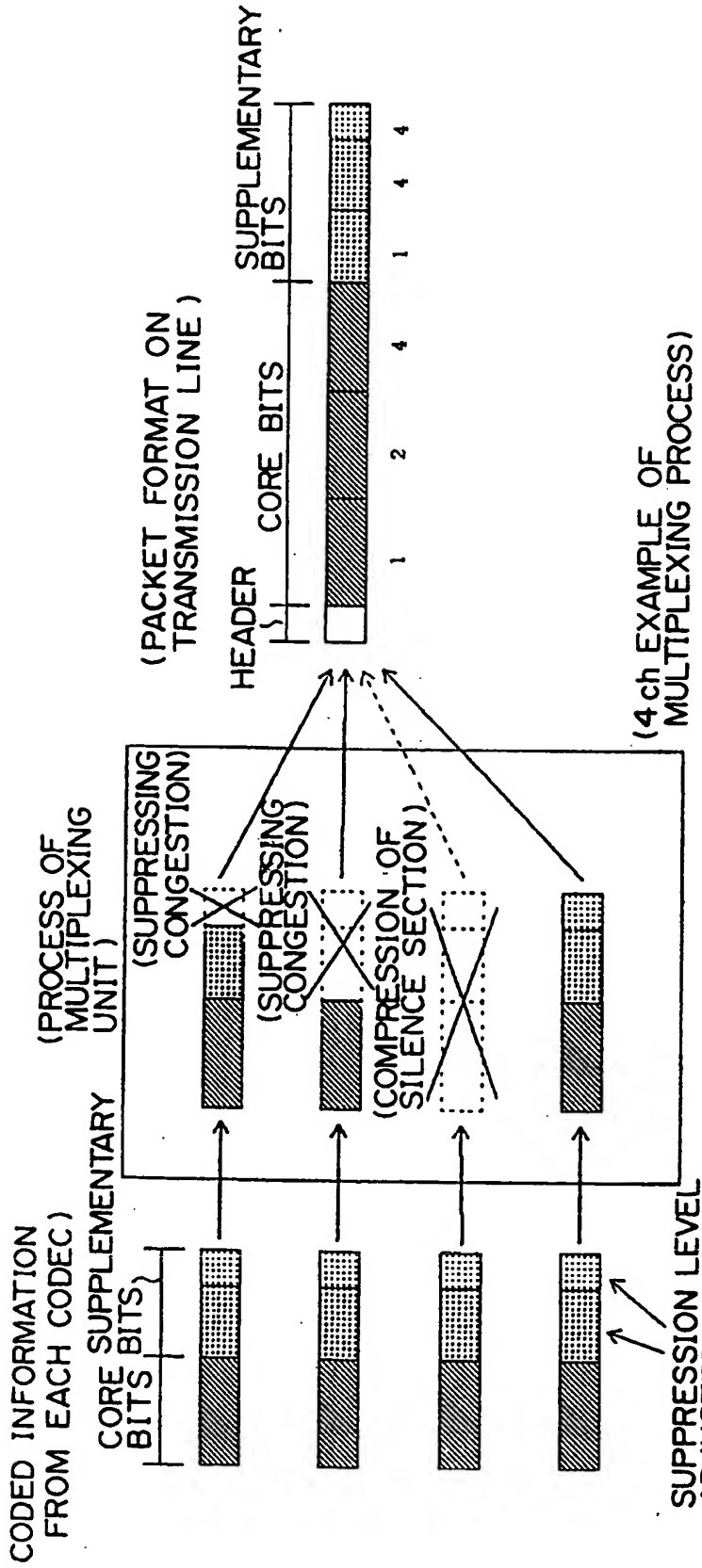

(CODED INFORMATION FROM EACH CODEC)
CORE BITS
SUPPLEMENTARY BITS
(PROCESS OF MULTIPLEXING UNIT)
(SUPPRESSING CONGESTION)
(SUPPRESSING CONGESTION)
(COMPRESSION OF SILENCE SECTION)
SUPPRESSION LEVEL ADJUSTED ACCORDING TO CONGESTION STATE
(PACKET FORMAT ON TRANSMISSION LINE)
HEADER
CORE BITS
SUPPLEMENTARY BITS
1
2
4
1
4
4
(4 ch EXAMPLE OF MULTIPLEXING PROCESS)

Fig. 16B

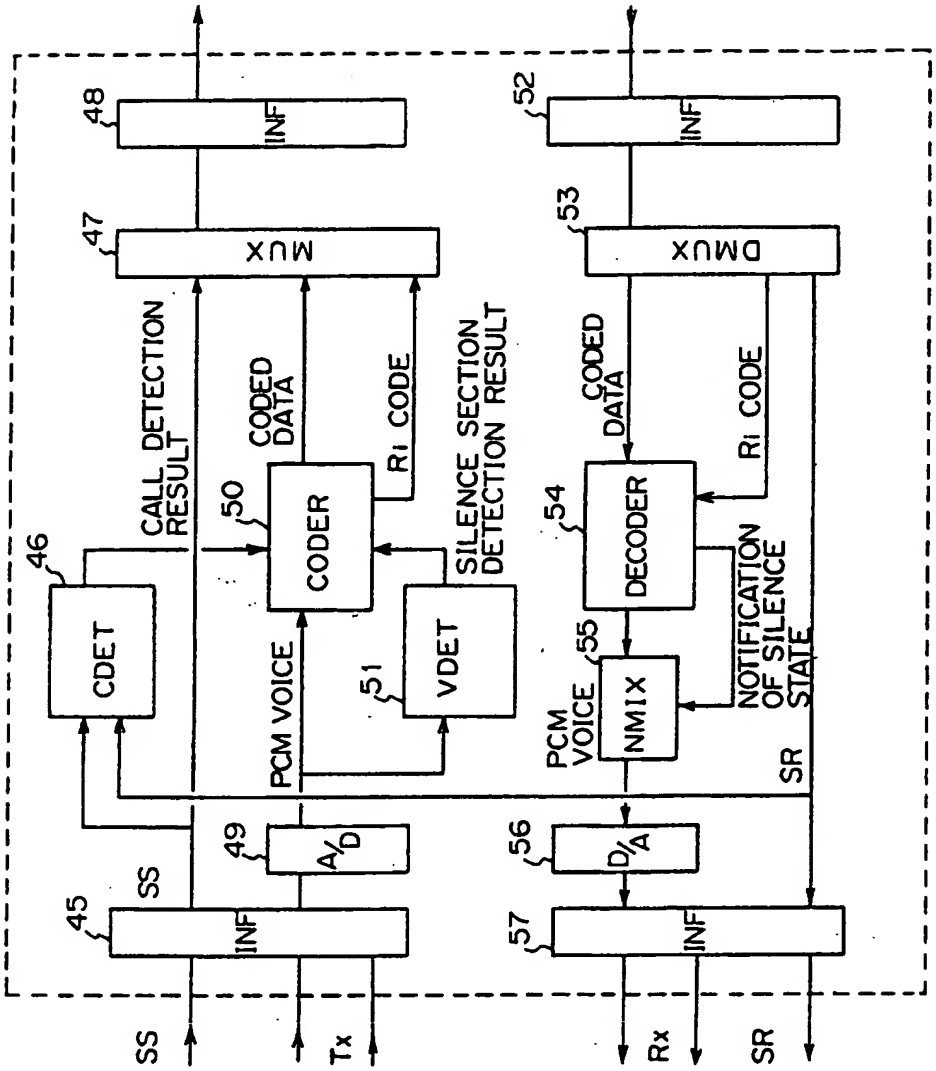

48
INF
47
MUX
52
INF
53
DMUX
CALL DETECTION RESULT
CODED DATA
Ri CODE
SILENCE SECTION DETECTION RESULT
CODED DATA
Ri CODE
46
CDET
50
CODER
54
DECODER
51
VDET
55
NMIX
PCM VOICE
NOTIFICATION OF SILENCE
PCM VOICE
SR STATE
49
A/D
56
D/A
SS
45
INF
57
INF
SS
Tx
Rx
SR

Fig. 17

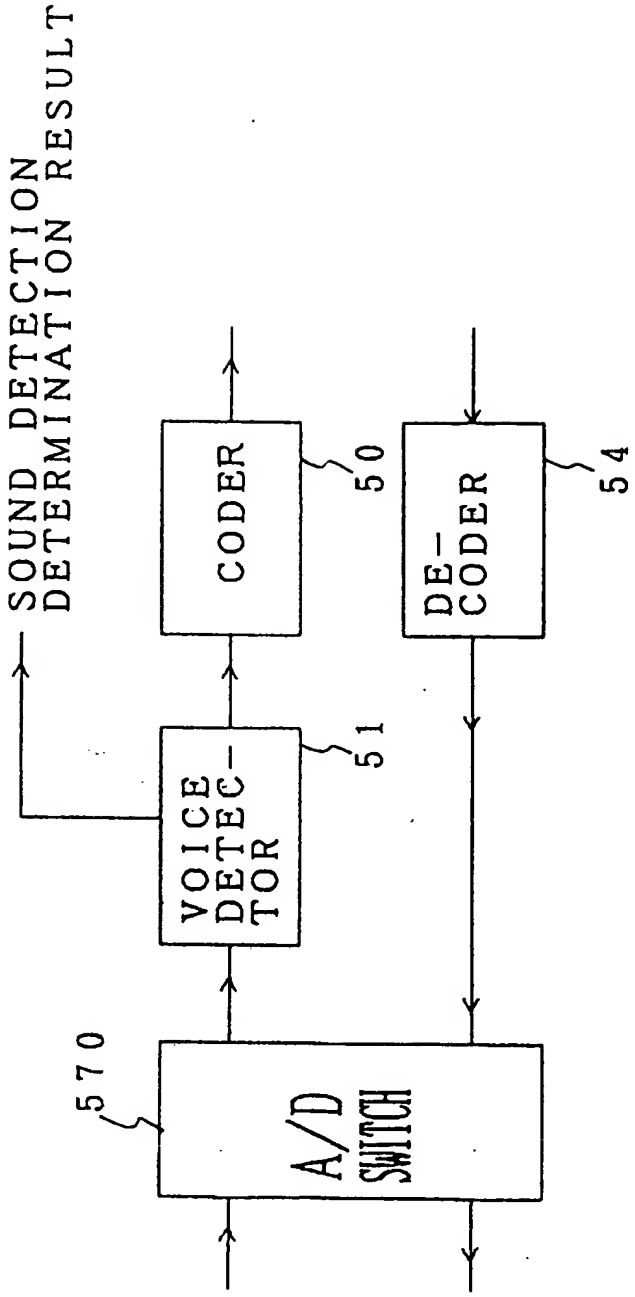

SOUND DETECTION DETERMINATION RESULT
CODER
50
DE-CODER
54
VOICE DETEC-TOR
51
A/D SWITCH
570

Fig. 18A

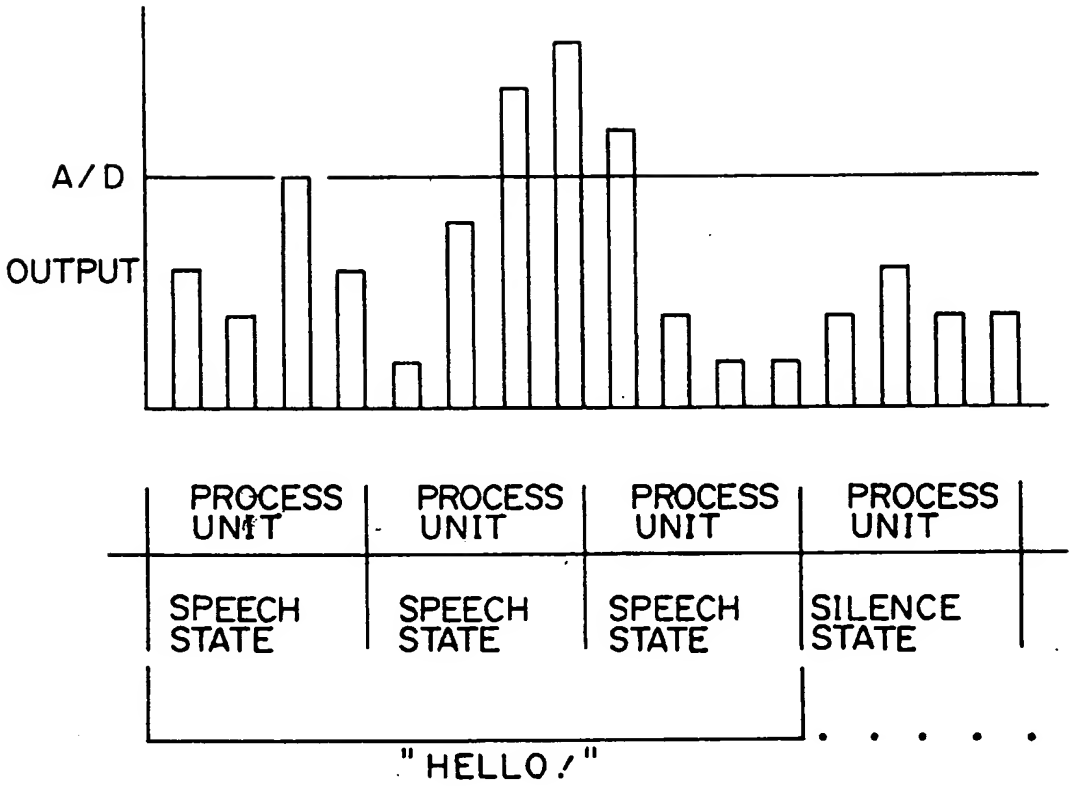

A/D
OUTPUT
PROCESS UNIT
PROCESS UNIT
PROCESS UNIT
PROCESS UNIT
SPEECH STATE
SPEECH STATE
SPEECH STATE
SILENCE STATE
"HELLO!"

Fig. 18B

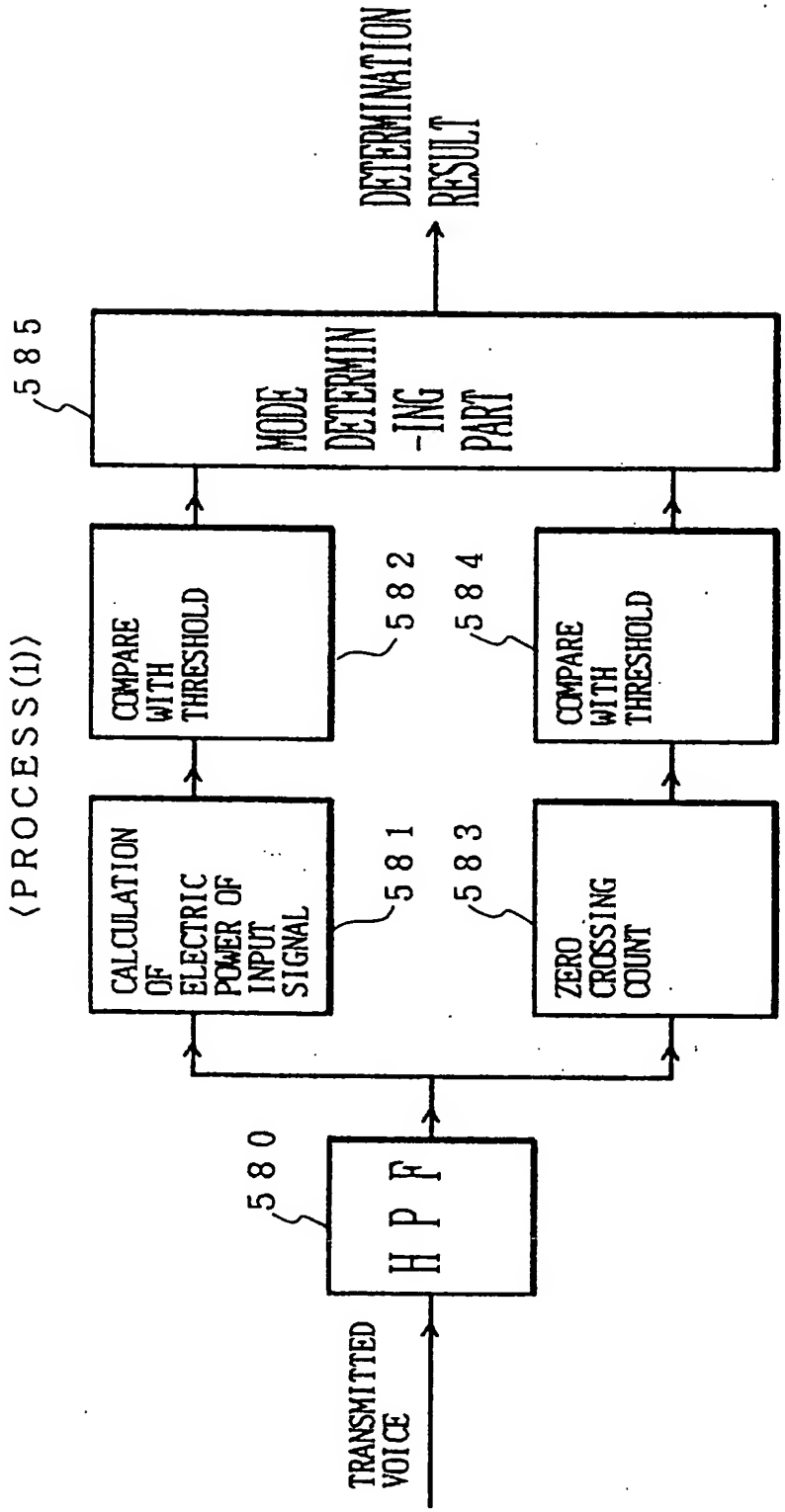

TRANSMITTED VOICE
580
HPF
581
CALCULATION OF ELECTRIC POWER OF INPUT SIGNAL
582
COMPARE WITH THRESHOLD
583
ZERO CROSSING COUNT
584
COMPARE WITH THRESHOLD
585
MODE DETERMIN -ING PART
DETERMINATION RESULT
〈PROCESS(1)〉
〈PROCESS(2)〉

Fig. 19

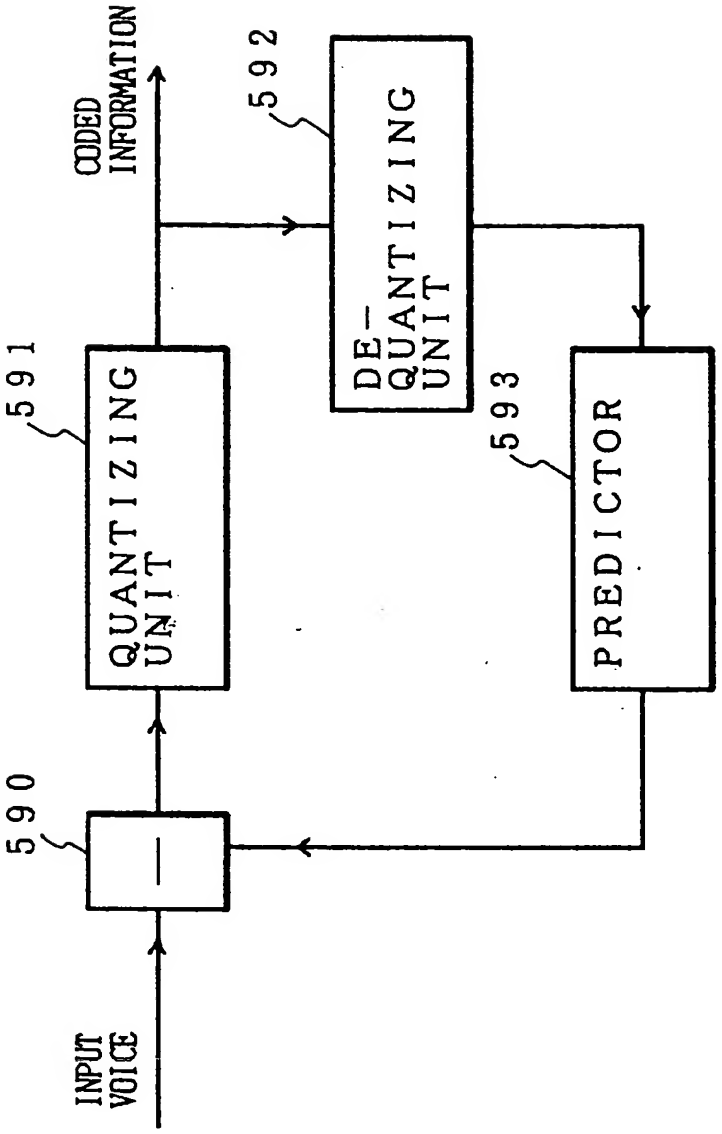

INPUT VOICE
590
591
QUANTIZING UNIT
CODED INFORMATION
592
DE-QUANTIZING UNIT
593
PREDICTOR

Fig. 20A

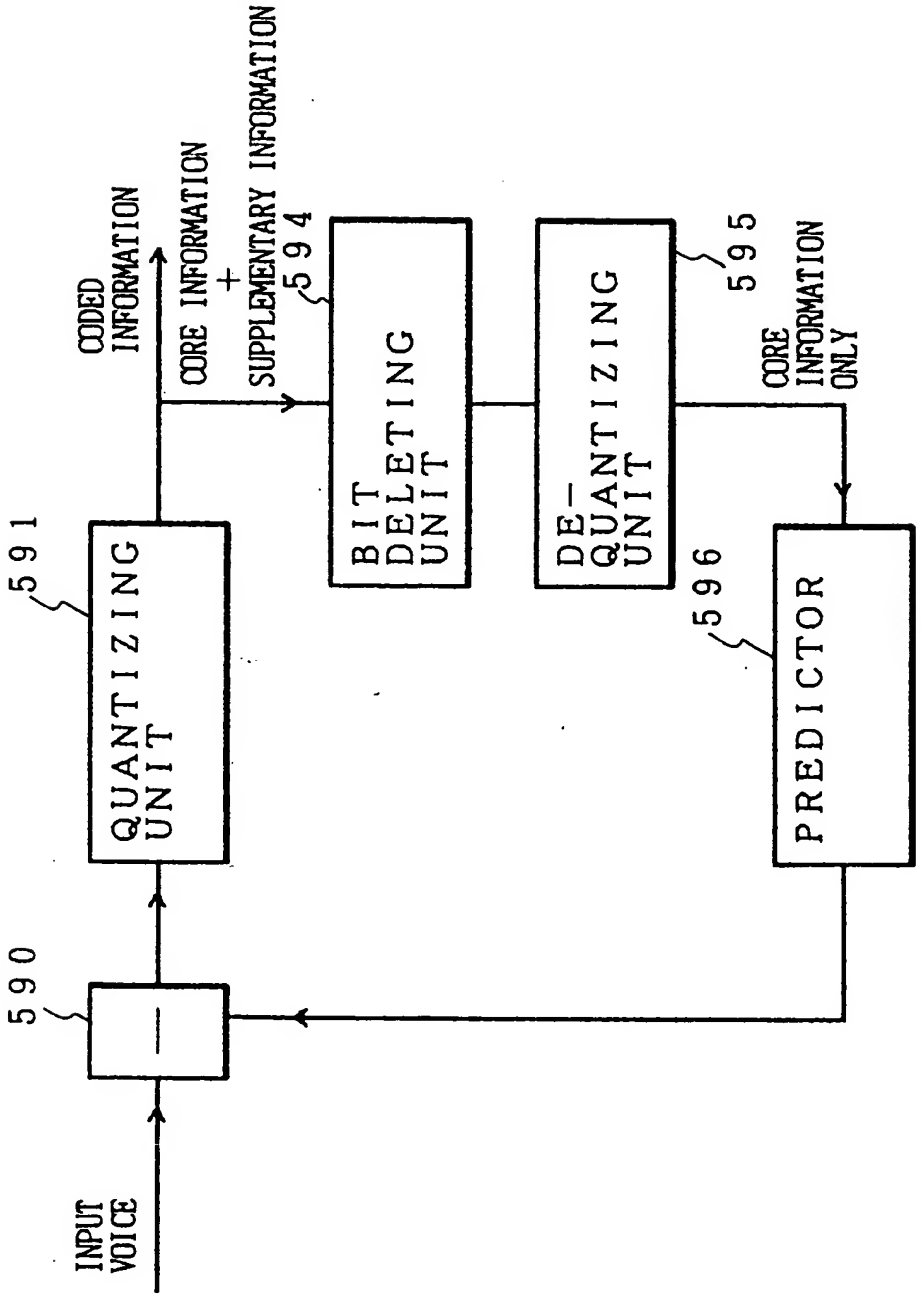

INPUT VOICE
590
591
QUANTIZING UNIT
CODED INFORMATION
CORE INFORMATION + SUPPLEMENTARY INFORMATION
594
BIT DELETING UNIT
DE-QUANTIZING UNIT
595
CORE INFORMATION ONLY
596
PREDICTOR

Fig. 20B

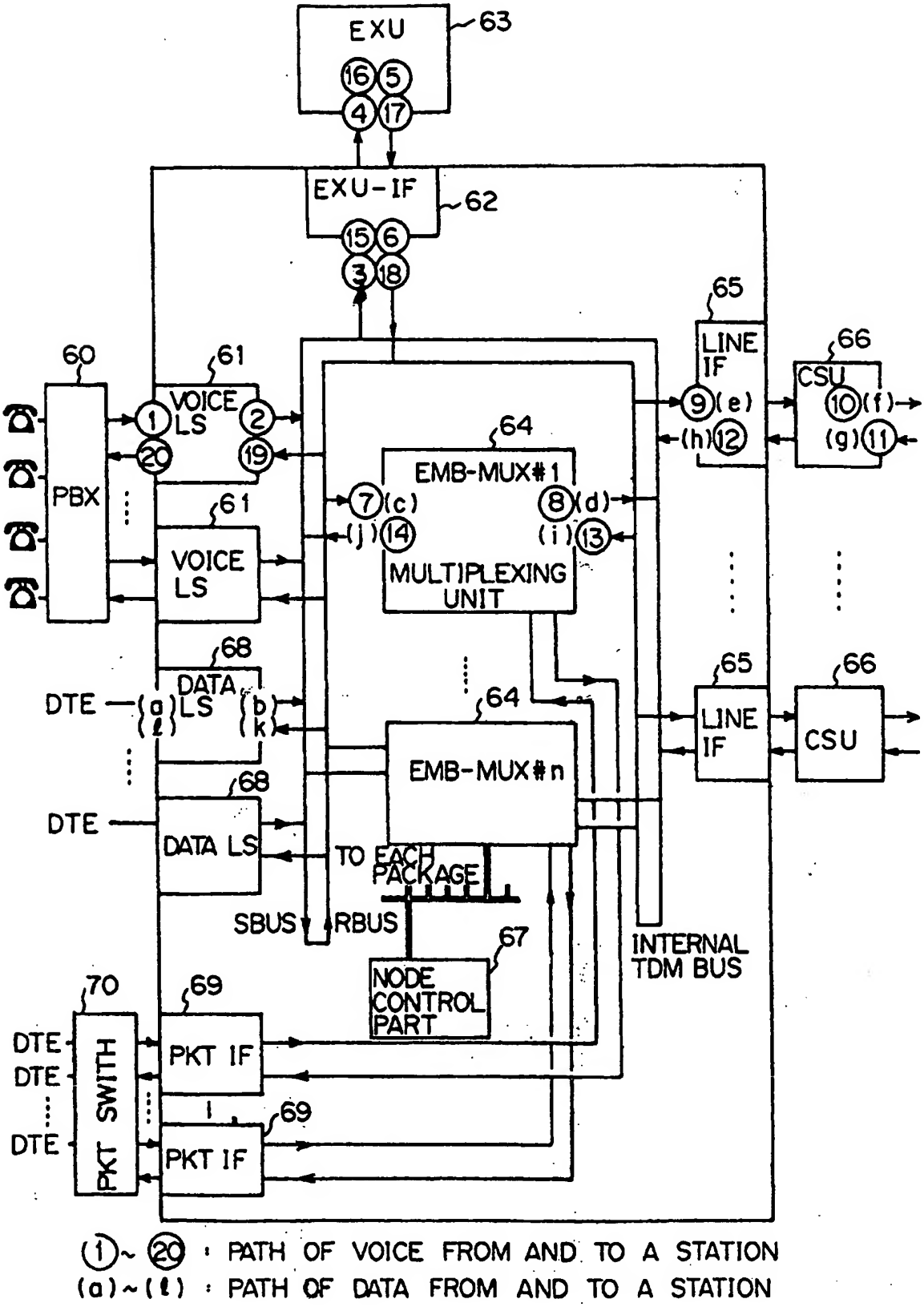

EXU
63
16
5
4
17
EXU-IF
62
15
6
3
18
65
LINE IF
66
CSU
60
61
VOICE LS
1
2
20
19
9
(e)
(h)
12
10
(f)
(g)
11
64
EMB-MUX#1
PBX
7
(c)
(j)
14
8
(d)
(i)
13
MULTIPLEXING UNIT
61
VOICE LS
68
DATA
DTE
(a)
(ℓ)
LS
b
k
65
66
64
LINE IF
CSU
EMB-MUX#n
68
DTE
DATA LS
TO EACH PACKAGE
SBUS
RBUS
67
INTERNAL TDM BUS
NODE CONTROL PART
70
69
DTE
DTE
DTE
PKT SWITH
PKT IF
PKT IF
69
(1) ~ (20) : PATH OF VOICE FROM AND TO A STATION
(a) ~ (ℓ) : PATH OF DATA FROM AND TO A STATION

Fig. 21

		b 3	b 2	b 1	b 0
VOICE CHANNEL		SIGNALING TRANS-MISSION FLAG	TRANSMISSION BAND INFORMATION (R I 2~R I 0)		
DATA CHANNEL	WITH Rs	R S SIGNAL TRANSMISSION FLAG	– (P K T-DATA)	– (P K T-DATA)	IDLE FLAG

Fig. 22

EP 0 464 839 A2

ITEM	OPERATION MODE			BETWEEN MUX AND DMUX
	MODE			RI2～0
0	NOT IN USE／ CALL DISCONNECTED		0	000
1	SILENCE MODE	LEVEL 1	2.0kbps	001
2	SPEECH MODE	LEVEL 2	6.0kbps	010
3		LEVEL 3	8.0kbps	011
4		LEVEL 4	10.0kbps	100
5		LEVEL 5	12.0kbps	101
6		LEVEL 6	14.0kbps	110
7		LEVEL 7	16.0kbps	111

Fig. 23

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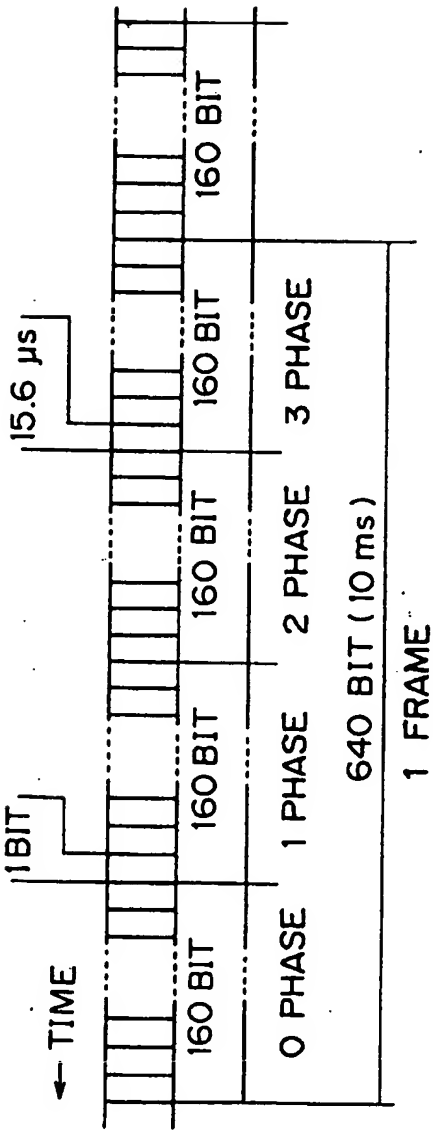
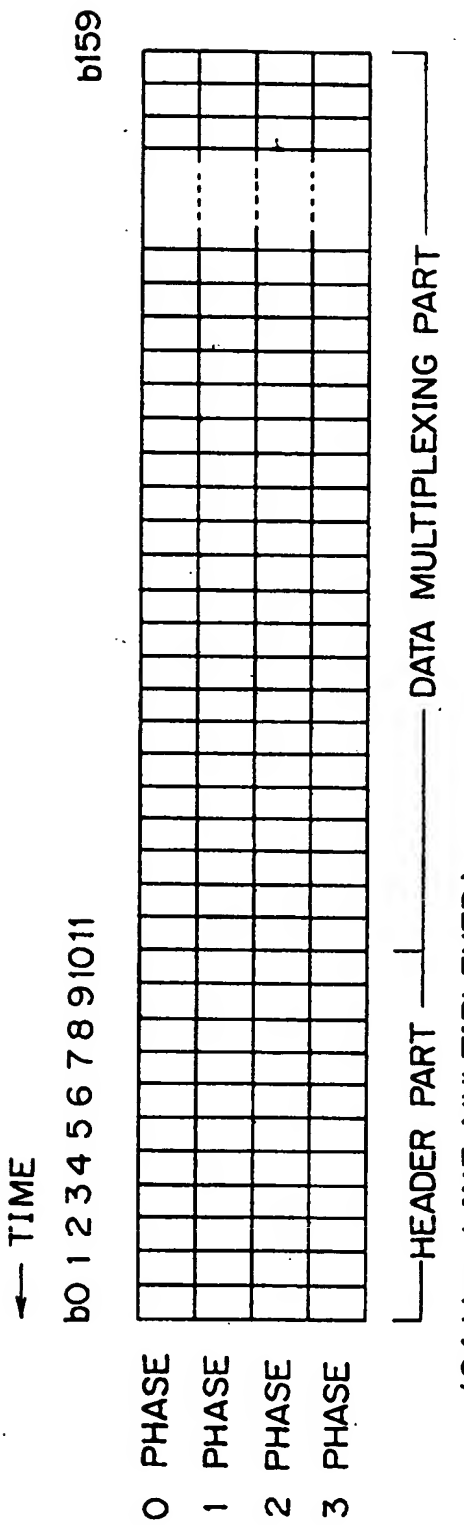

← TIME
1 BIT
15.6 μs
160 BIT
160 BIT
160 BIT
160 BIT
0 PHASE
1 PHASE
2 PHASE
3 PHASE
640 BIT (10 ms)
1 FRAME

Fig. 24



HEADER PART : LEADING 11 BITS OF EACH OF 0 ~ 3 PHASE (TOTAL 44 BITS)
DATA MULTIPLEXING PART : BITS FOLLOWING AND INCLUDING 12 TH BITS OF EACH OF 0 ~ 3 PHASE (TOTAL 596 BITS)

Fig. 25A

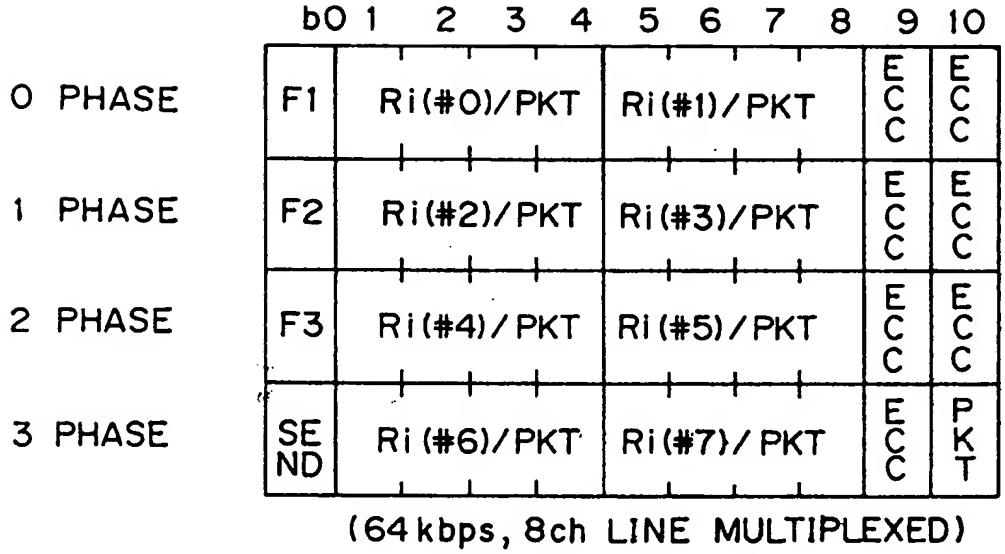

b0 1 2 3 4 5 6 7 8 9 10
0 PHASE
1 PHASE
2 PHASE
3 PHASE
F1
Ri(#0)/PKT
Ri(#1)/PKT
ECC
ECC
F2
Ri(#2)/PKT
Ri(#3)/PKT
ECC
ECC
F3
Ri(#4)/PKT
Ri(#5)/PKT
ECC
ECC
SEND
Ri(#6)/PKT
Ri(#7)/PKT
ECC
PKT
(64kbps, 8ch LINE MULTIPLEXED)

Fig. 25B

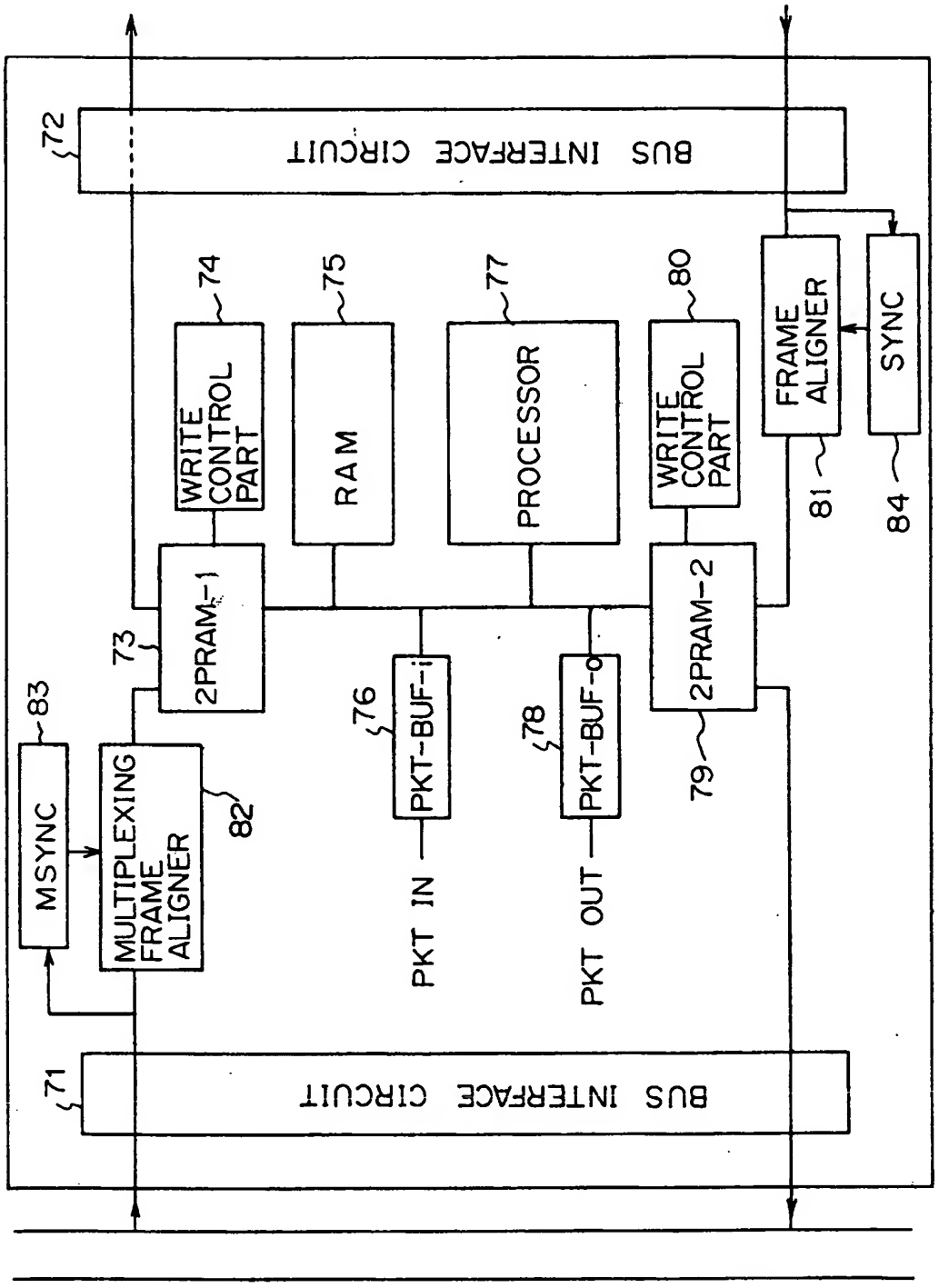

BUS INTERFACE CIRCUIT
72
WRITE CONTROL PART
74
RAM
75
PROCESSOR
77
WRITE CONTROL PART
80
FRAME ALIGNER
81
SYNC
84
2PRAM-1
73
2PRAM-2
79
PKT-BUF-i
76
PKT-BUF-o
78
PKT IN
PKT OUT
MSYNC
83
MULTIPLEXING FRAME ALIGNER
82
BUS INTERFACE CIRCUIT
71

Fig. 26

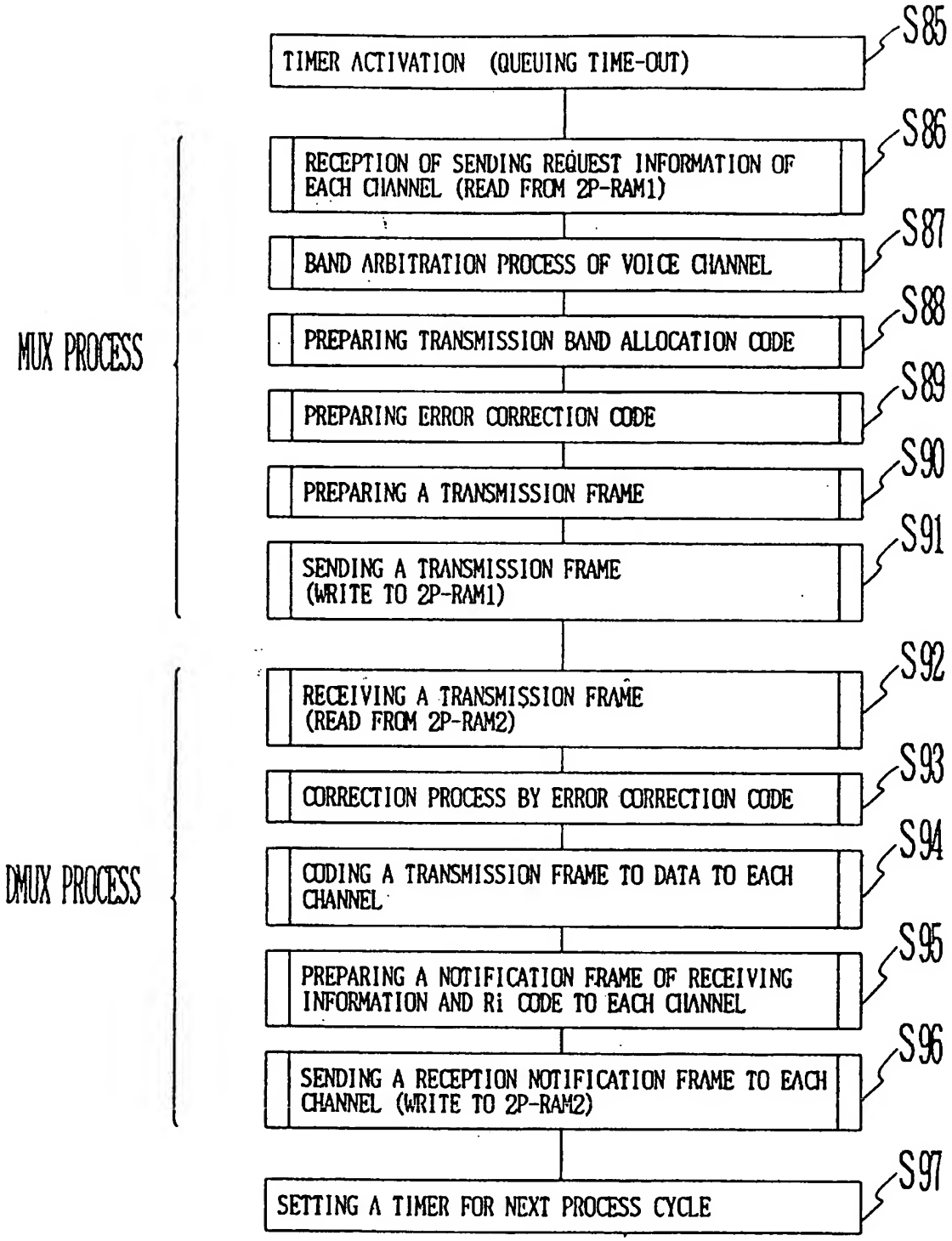
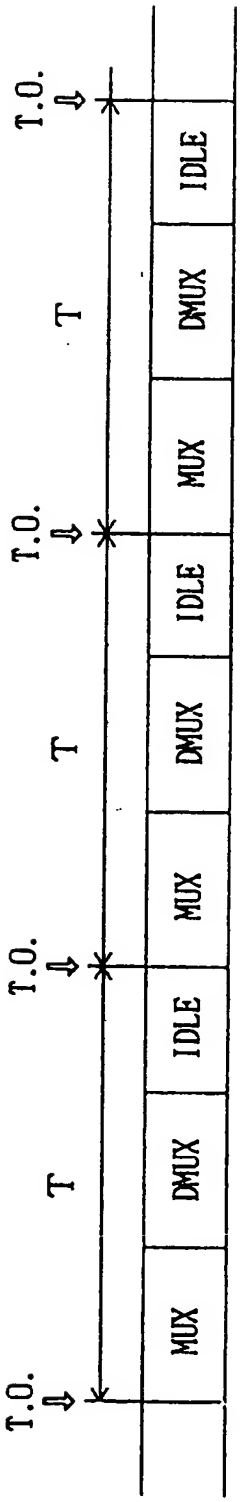

MUX PROCESS
DMUX PROCESS
TIMER ACTIVATION (QUEUING TIME-OUT)
S85
RECEPTION OF SENDING REQUEST INFORMATION OF EACH CHANNEL (READ FROM 2P-RAM1)
S86
BAND ARBITRATION PROCESS OF VOICE CHANNEL
S87
PREPARING TRANSMISSION BAND ALLOCATION CODE
S88
PREPARING ERROR CORRECTION CODE
S89
PREPARING A TRANSMISSION FRAME
S90
SENDING A TRANSMISSION FRAME (WRITE TO 2P-RAM1)
S91
RECEIVING A TRANSMISSION FRAME (READ FROM 2P-RAM2)
S92
CORRECTION PROCESS BY ERROR CORRECTION CODE
S93
CODING A TRANSMISSION FRAME TO DATA TO EACH CHANNEL
S94
PREPARING A NOTIFICATION FRAME OF RECEIVING INFORMATION AND Ri CODE TO EACH CHANNEL
S95
SENDING A RECEPTION NOTIFICATION FRAME TO EACH CHANNEL (WRITE TO 2P-RAM2)
S96
SETTING A TIMER FOR NEXT PROCESS CYCLE
S97

Fig. 27



WHERE : T.O. : TIME-OUT OF A TIMER
IDLE : TIME-OUT WAITING MODE OF A TIMER

Fig. 28

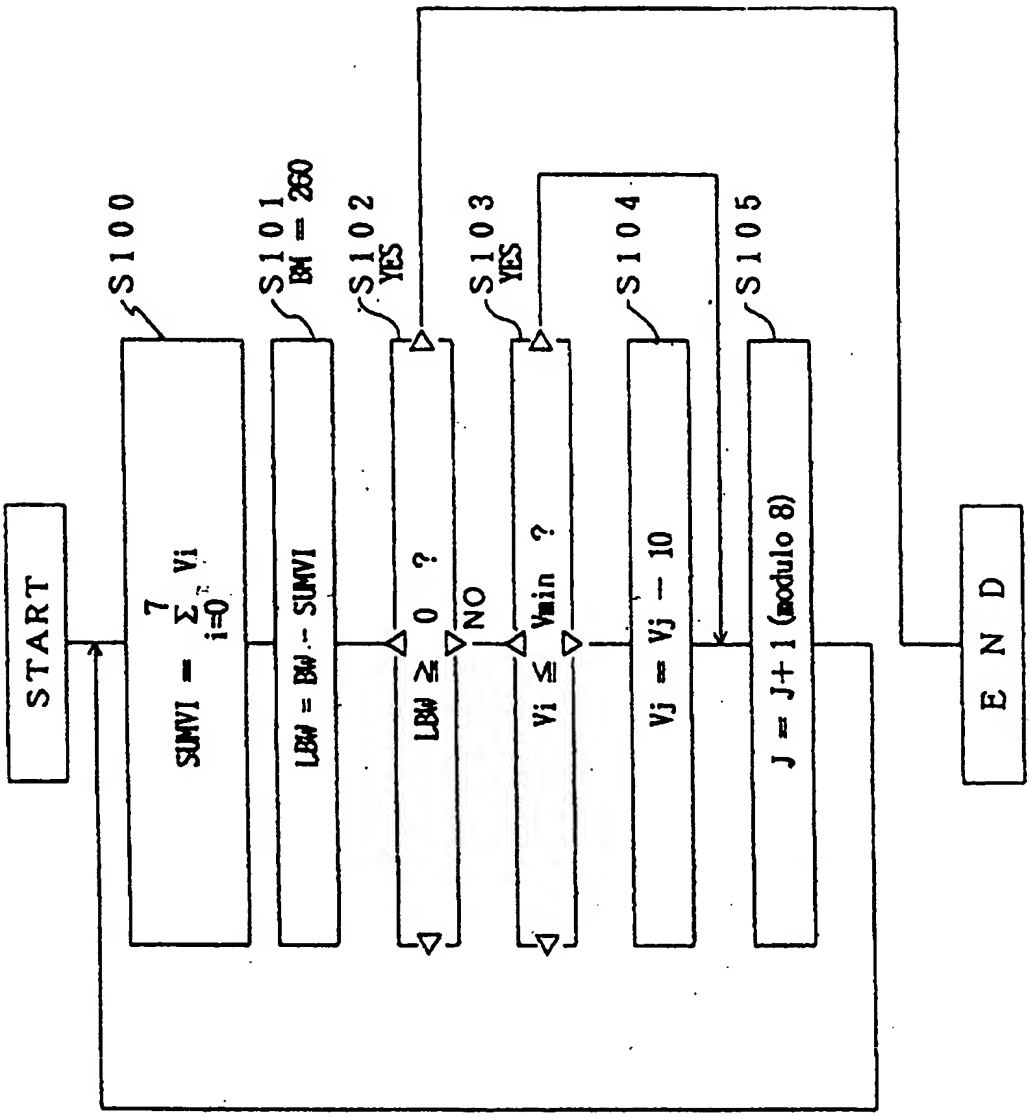

START
S100
SUMVI = Σ Vi (i=0 to 7)
S101
LBW = BW - SUMVI
S102
LBW ≧ 0 ?
YES
NO
S103
Vi ≦ Vmin ?
YES
S104
Vj = Vj - 10
S105
J = J+1 (modulo 8)
END

Fig. 29

	CH−1	CH−2	CH−3	CH−4	CH−5	CH−6	CH−7	CH−8	BW	REQUEST ↓	ALLOCATION ↓
T= 1	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	: 260 ;	80	/ 80
T= 2	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 40	50 / 40	: 260 ;	400	/260
T= 3	70 / 30	70 / 30	70 / 30	70 / 30	70 / 40	70 / 40	70 / 30	70 / 30	: 260 ;	560	/260
T= 4	80 / 30	80 / 30	80 / 40	80 / 40	80 / 30	80 / 30	80 / 30	80 / 30	: 260 ;	640	/260
T= 5	80 / 40	80 / 40	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	: 260 ;	640	/260
T= 6	70 / 30	70 / 30	70 / 30	70 / 30	70 / 30	70 / 40	70 / 40	60 / 30	: 260 ;	550	/260
T= 7	80 / 30	80 / 30	80 / 30	80 / 40	80 / 40	80 / 30	80 / 30	40 / 30	: 260 ;	600	/260
T= 8	60 / 30	60 / 40	60 / 40	60 / 30	60 / 30	60 / 30	60 / 30	30 / 30	: 260 ;	450	/260
T= 9	50 / 40	50 / 30	50 / 30	50 / 30	50 / 40	50 / 40	50 / 40	10 / 10	: 260 ;	360	/260
T=10	30 / 30	30 / 30	30 / 30	30 / 30	30 / 30	30 / 30	30 / 30	10 / 10	: 260 ;	220	/220
T=11	40 / 40	40 / 40	10 / 10	40 / 40	10 / 10	10 / 10	10 / 10	30 / 30	: 260 ;	190	/190
T=12	50 / 50	50 / 50	10 / 10	50 / 50	10 / 10	10 / 10	10 / 10	30 / 30	: 260 ;	220	/220
T=13	80 / 70	80 / 70	10 / 10	60 / 50	10 / 10	10 / 10	10 / 10	30 / 30	: 260 ;	290	/260
T=14	60 / 40	60 / 40	10 / 10	80 / 70	30 / 30	50 / 30	10 / 10	50 / 30	: 260 ;	350	/260
T=15	50 / 40	50 / 40	0 / 0	70 / 50	40 / 30	80 / 60	10 / 10	50 / 30	: 260 ;	350	/260

Fig. 30

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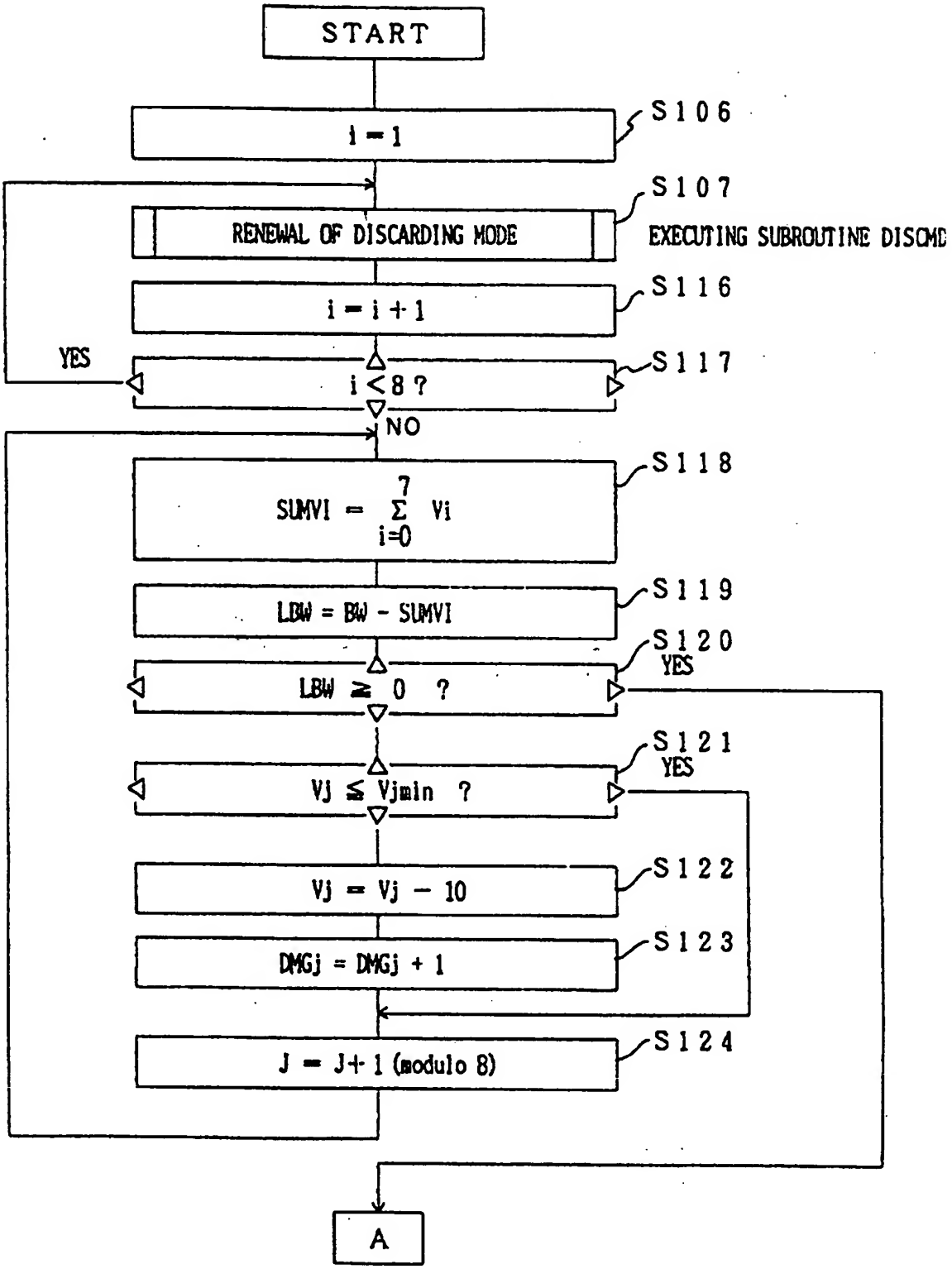

START
i = 1
S106
RENEWAL OF DISCARDING MODE
S107
EXECUTING SUBROUTINE DISCMD
i = i + 1
S116
YES
i ≤ 8 ?
S117
NO
SUMVI = Σ Vi (i=0 to 7)
S118
LBW = BW - SUMVI
S119
LBW ≧ 0 ?
S120
YES
Vj ≦ Vjmin ?
S121
YES
Vj = Vj - 10
S122
DMGj = DMGj + 1
S123
J = J+1 (modulo 8)
S124
A

Fig. 31A

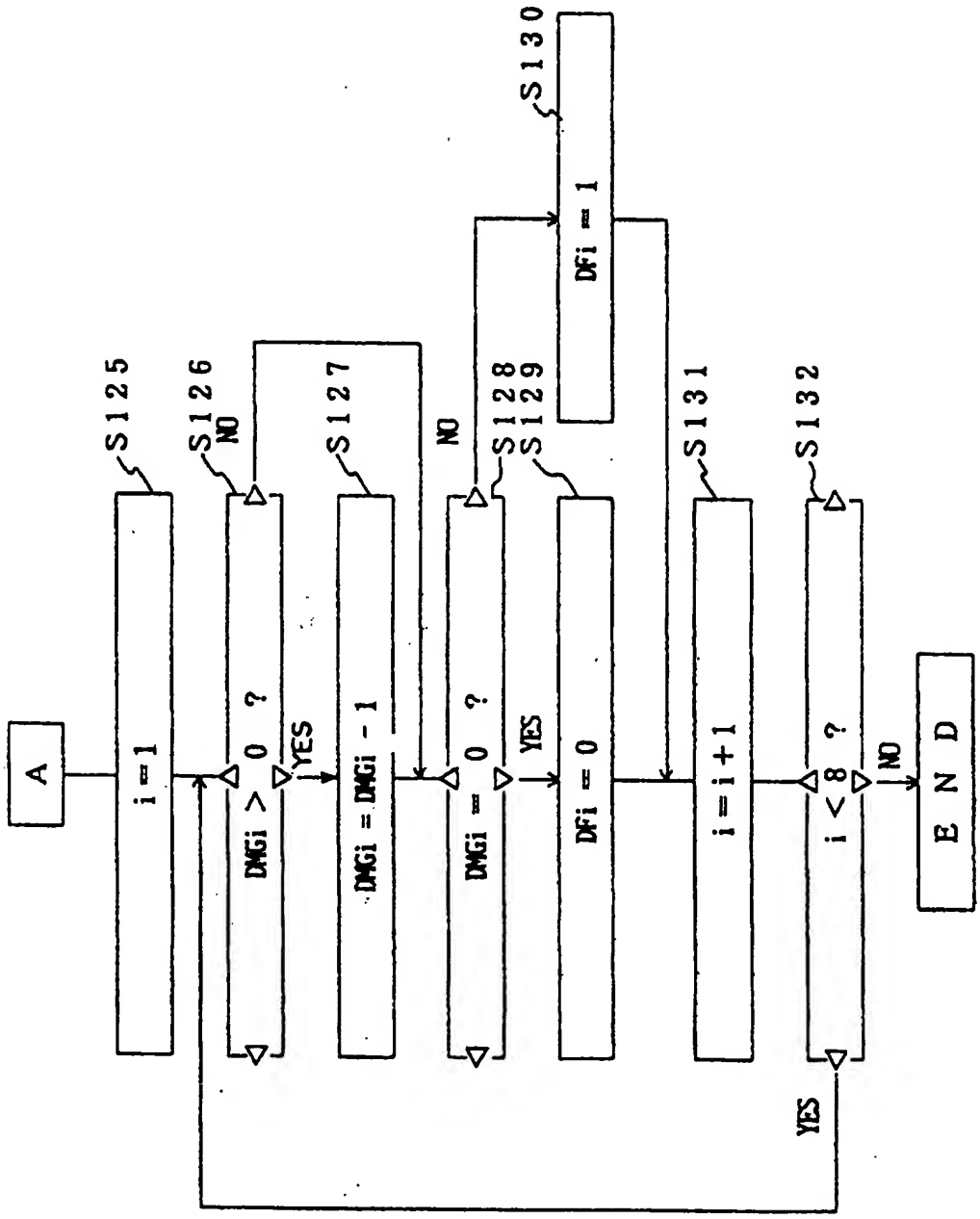

A
i = 1
S125
DMGi > 0 ?
S126
NO
YES
DMGi = DMGi - 1
S127
DMGi = 0 ?
NO
S128
S129
YES
DFi = 0
DFi = 1
S130
i = i + 1
S131
i < 8 ?
S132
NO
YES
E N D

Fig. 31B

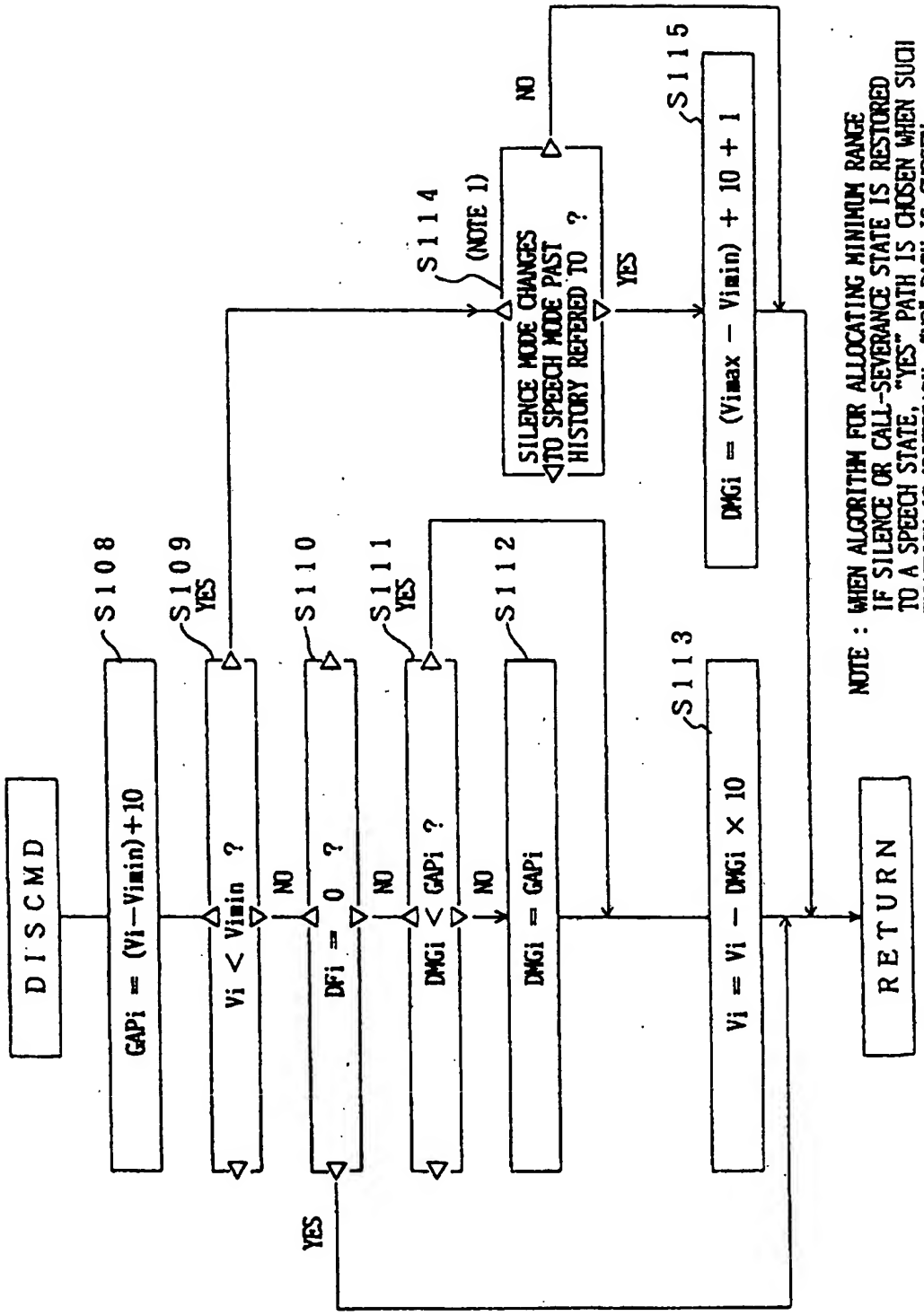

DISCMD
S108
GAPi = (Vi−Vimin)+10
S109
Vi < Vimin ?
YES
NO
S110
DFi = 0 ?
YES
NO
S111
DMGi < GAPi ?
YES
NO
S112
DMGi = GAPi
S113
Vi = Vi − DMGi × 10
S114
(NOTE 1)
SILENCE MODE CHANGES TO SPEECH MODE PAST HISTORY REFERED TO ?
YES
NO
S115
DMGi = (Vimax − Vimin) + 10 + 1
RETURN
NOTE : WHEN ALGORITHM FOR ALLOCATING MINIMUM RANGE IF SILENCE OR CALL-SEVERANCE STATE IS RESTORED TO A SPEECH STATE, "YES" PATH IS CHOSEN WHEN SUCH FUNCTION IS UNNESSARY, "NO" PATH IS CHOSEN

Fig. 32

	CH−1	CH−2	CH−3	CH−4	CH−5	CH−6	CH−7	CH−8	BW	REQUEST	ALLOCATION
T= 1	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	10 / 10	80	80	80
	1 /5 /9	1 /5 /9	1 /5 /9	1 /5 /9	1 /5 /9	1 /5 /9	1 /5 /9	1 /5 /9			
T= 2	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	240	400	240
	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2			
T= 3	70 / 30	70 / 30	70 / 30	70 / 30	70 / 30	70 / 30	70 / 40	70 / 40	480	560	260
	1 /3 /4	1 /3 /4	1 /3 /4	1 /3 /4	1 /3 /4	1 /3 /4	1 /2 /4	1 /2 /4			
T= 4	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	80 / 40	80 / 40	420	640	260
	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /3 /5	1 /3 /5			
T= 5	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	80 / 40	80 / 40	340	640	260
	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /3 /5	1 /3 /5			
T= 6	70 / 30	70 / 30	70 / 30	70 / 30	70 / 30	70 / 30	70 / 40	60 / 30	250	550	250
	1 /3 /4	1 /3 /4	1 /3 /4	1 /3 /4	1 /3 /4	1 /3 /4	1 /2 /4	1 /2 /3			
T= 7	80 / 30	80 / 30	80 / 30	80 / 30	80 / 30	80 / 40	80 / 40	40 / 30	390	600	260
	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /4 /5	1 /3 /5	1 /3 /5	0 /0 /1			
T= 8	60 / 30	60 / 30	60 / 30	60 / 30	60 / 30	60 / 30	60 / 30	30 / 30	240	450	240
	1 /2 /3	1 /2 /3	1 /2 /3	1 /2 /3	1 /2 /3	1 /2 /3	1 /2 /3	0 /0 /0			
T= 9	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	50 / 30	10 / 10	220	360	220
	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /1 /2	1 /5 /9			
T=10	30 / 30	30 / 30	30 / 30	30 / 30	30 / 30	30 / 30	30 / 30	10 / 10	220	220	220
	0 /0 /0	0 /0 /0	0 /0 /0	0 /0 /0	0 /0 /0	0 /0 /0	0 /0 /0	1 /5 /9			
T=11	40 / 40	40 / 40	10 / 10	40 / 40	10 / 10	10 / 10	10 / 10	30 / 30	190	190	190
	0 /0 /1	0 /0 /1	1 /5 /9	0 /0 /1	1 /5 /9	1 /5 /9	1 /5 /9	0 /0 /0			
T=12	50 / 50	50 / 50	10 / 10	50 / 50	10 / 10	10 / 10	10 / 10	30 / 30	220	220	220
	0 /0 /2	0 /0 /2	1 /5 /9	0 /0 /2	1 /5 /9	1 /5 /9	1 /5 /9	0 /0 /0			
T=13	80 / 70	80 / 70	10 / 10	60 / 50	10 / 10	10 / 10	10 / 10	30 / 30	290	290	260
	0 /0 /5	0 /0 /5	1 /5 /9	0 /0 /3	1 /5 /9	1 /5 /9	1 /5 /9	0 /0 /0			
T=14	60 / 40	60 / 40	10 / 10	80 / 70	30 / 30	50 / 30	10 / 10	50 / 30	330	350	260
	1 /1 /3	1 /1 /3	1 /5 /9	0 /0 /5	0 /0 /0	1 /1 /2	1 /5 /9	1 /1 /2			
T=15	50 / 30	50 / 40	0 / 0	70 / 60	40 / 30	80 / 60	10 / 10	50 / 30	310	350	260
	1 /1 /2	0 /0 /2	1 /5 /9	0 /0 /4	0 /0 /1	1 /1 /5	1 /5 /9	1 /1 /2			

Fig. 33A

TIME	T	= 1	2	3	4	5	6	7	8	9	10	11
REQUESTED BITS	Vi	=80	10	10	80	80	80	80	80	80	80	80
	GAPi	= 5	-2	-2	5	5	5	5	5	5	5	5
PRECEDING TIME	DMGi	=—	0	5	5	4	3	2	1	0	0	0
FOLLOWING TIME	DMGi	= 0	5	5	4	3	2	1	0	0	0	0
ALLOCATED BITS	Vi'	=80	10	10	30	40	50	60	70	80	80	80
DISCARDING FLAG	DF'	= 0	1	1	1	1	1	1	0	0	0	0

SILENCE MODE SECTION

DRAGGING DISCARDING HISTORY

Fig. 33B

EP 0 464 839 A2

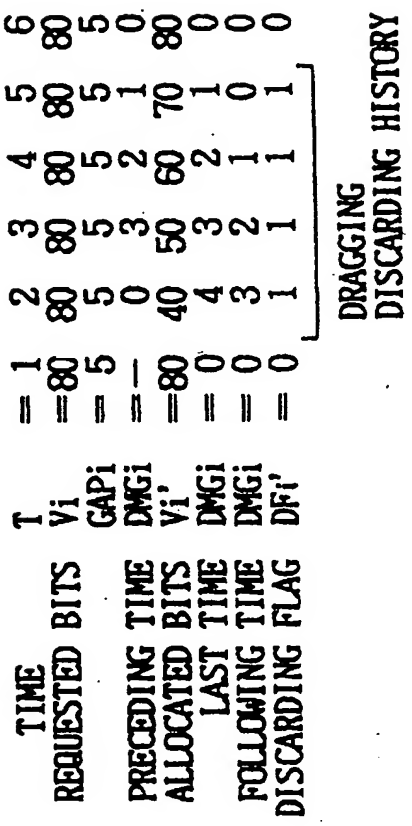
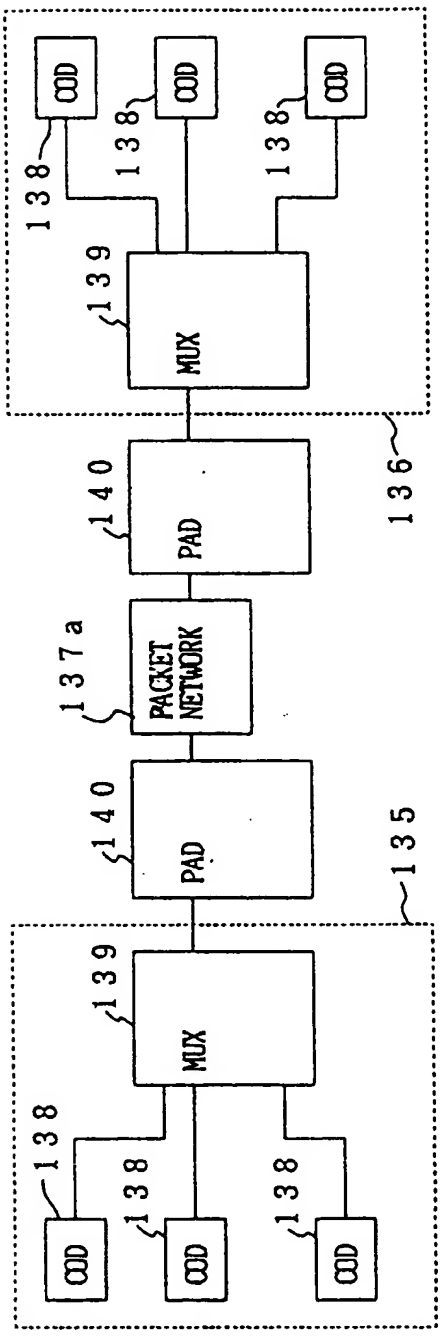

TIME
REQUESTED BITS
PRECEDING TIME
ALLOCATED BITS
LAST TIME
FOLLOWING TIME
DISCARDING FLAG
T
Vi
GAPi
DMGi
Vi'
DMGi
DMGi
DFi'
= 1 2 3 4 5 6
=80 80 80 80 80 80
= 5 5 5 5 5 5
= — 0 3 2 1 0
=80 40 50 60 70 80
= 0 4 3 2 1 0
= 0 3 2 1 0 0
= 0 1 1 1 1 0
DRAGGING
DISCARDING HISTORY

Fig. 33C

REQUEST Vi	40	40	50	60	70	80	40	50	30
PRECEDING TIME DMGi	0	2	2	2	2	2	1	1	1
GAPi	1	1	2	3	4	5	1	2	0
DMGi	0	1	2	2	2	2	1	1	0
NEW Vi	40	30	30	40	50	60	30	40	30
ACTUAL Vi	40	40	40	40	40	40	40	40	40
LAST TIME Vi	40	30	30	40	40(-10)	40(-20)	30	40	30
LAST TIME DMGi	0	1	2	2	3 (+1)	4 (+2)	1	1	0
FOLLOWING TIME DMGi	0	0	1	1	2	3	0	0	0
DFi	0	0	1	1	1	1	0	0	0

Fig. 34



COD : VOICE CODER (CODEC)
MUX : MULTIPLEXER
PAD : PKT INTERFACE PART (PACKET ASSEMBLER / DISSEMBLER)

Fig. 35

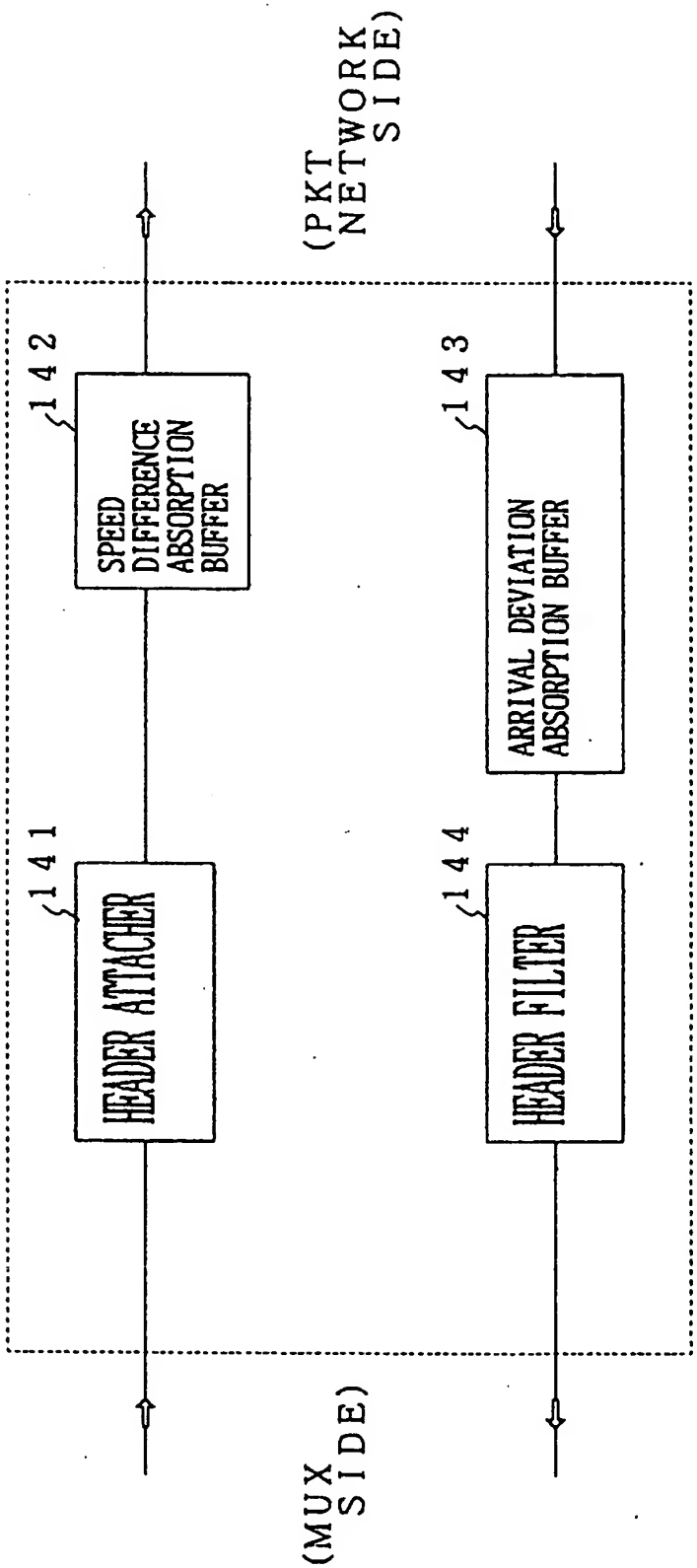

(PKT NETWORK SIDE)
142
SPEED DIFFERENCE ABSORPTION BUFFER
143
ARRIVAL DEVIATION ABSORPTION BUFFER
141
HEADER ATTACHER
144
HEADER FILTER
(MUX SIDE)

Fig. 36

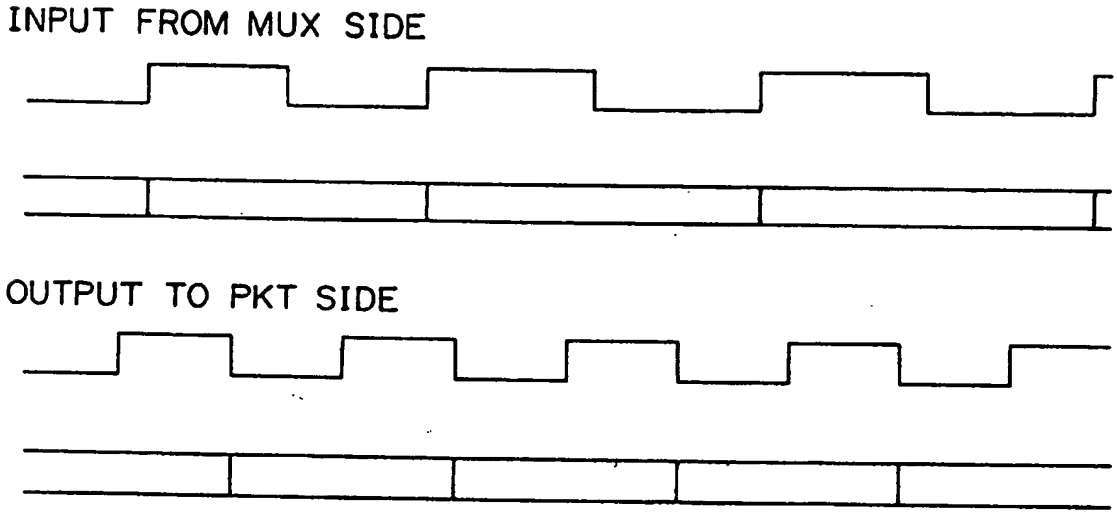


Fig. 37

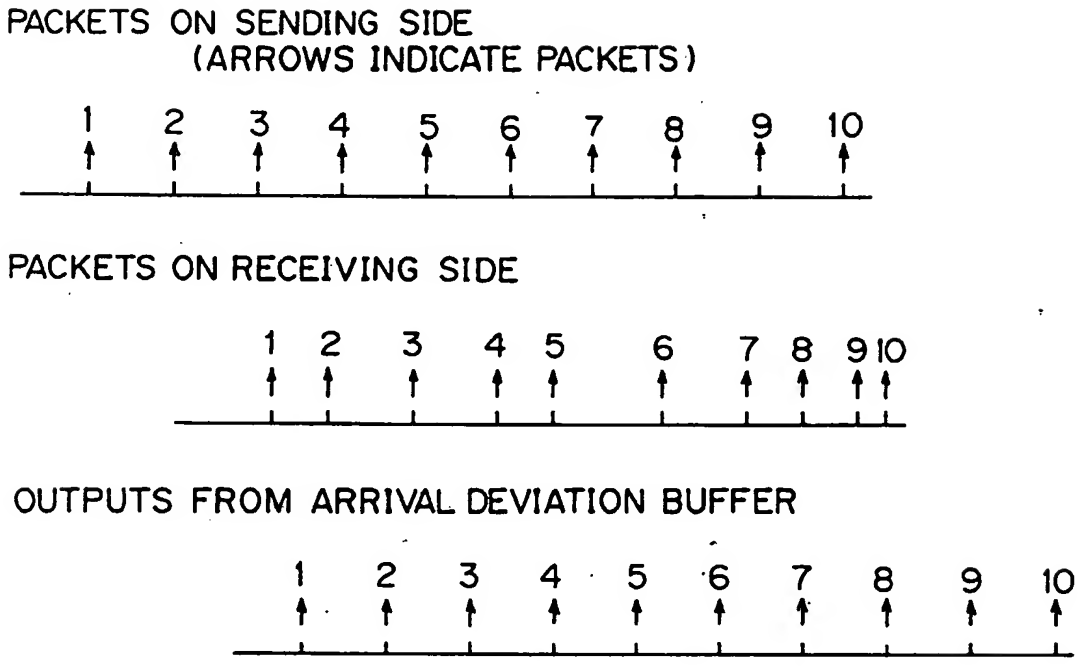

PACKETS ON SENDING SIDE
(ARROWS INDICATE PACKETS)
1 2 3 4 5 6 7 8 9 10
PACKETS ON RECEIVING SIDE
1 2 3 4 5 6 7 8 9 10
OUTPUTS FROM ARRIVAL DEVIATION BUFFER
1 2 3 4 5 6 7 8 9 10

Fig. 38

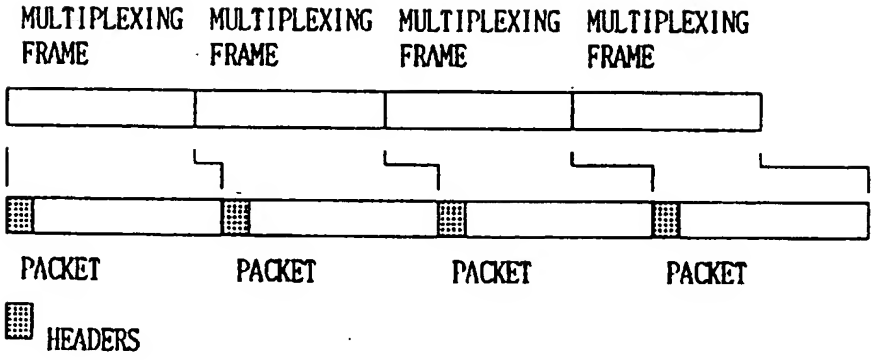


Fig. 39A

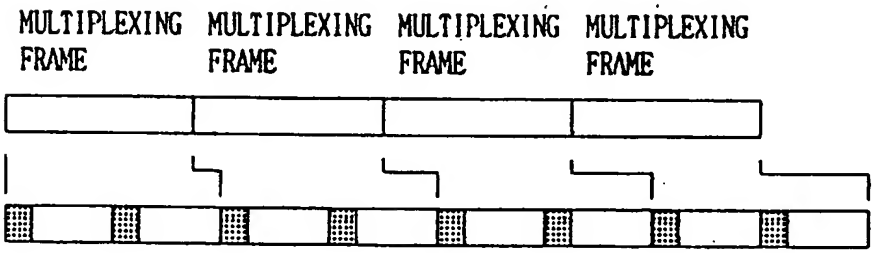


Fig. 39B

	OPERATING MODE				BETWEEN MUX AND DMUX R I 2～0
			(S,B1,B2,B3,B4)		
(1)	NOT IN USE		: (0,0,0,0,0)	0kbps	000
(2)	SILENCE TIME		: (0,0,0,0,0)	0kbps	001
(3)	SPEECH TIME	DISCARDING LEVEL 5	: (0,2,2,2,0)	12kbps	010
(4)		DISCARDING LEVEL 4	: (0,2,2,2,2)	16kbps	011
(5)		DISCARDING LEVEL 3	: (0,4,3,3,3)	26kbps	100
(6)		DISCARDING LEVEL 2	: (0,5,5,4,4)	36kbps	101
(7)		DISCARDING LEVEL 1	: (0,5,5,5,5)	40kbps	110

S : SIDE INFORMATION

Fig. 40

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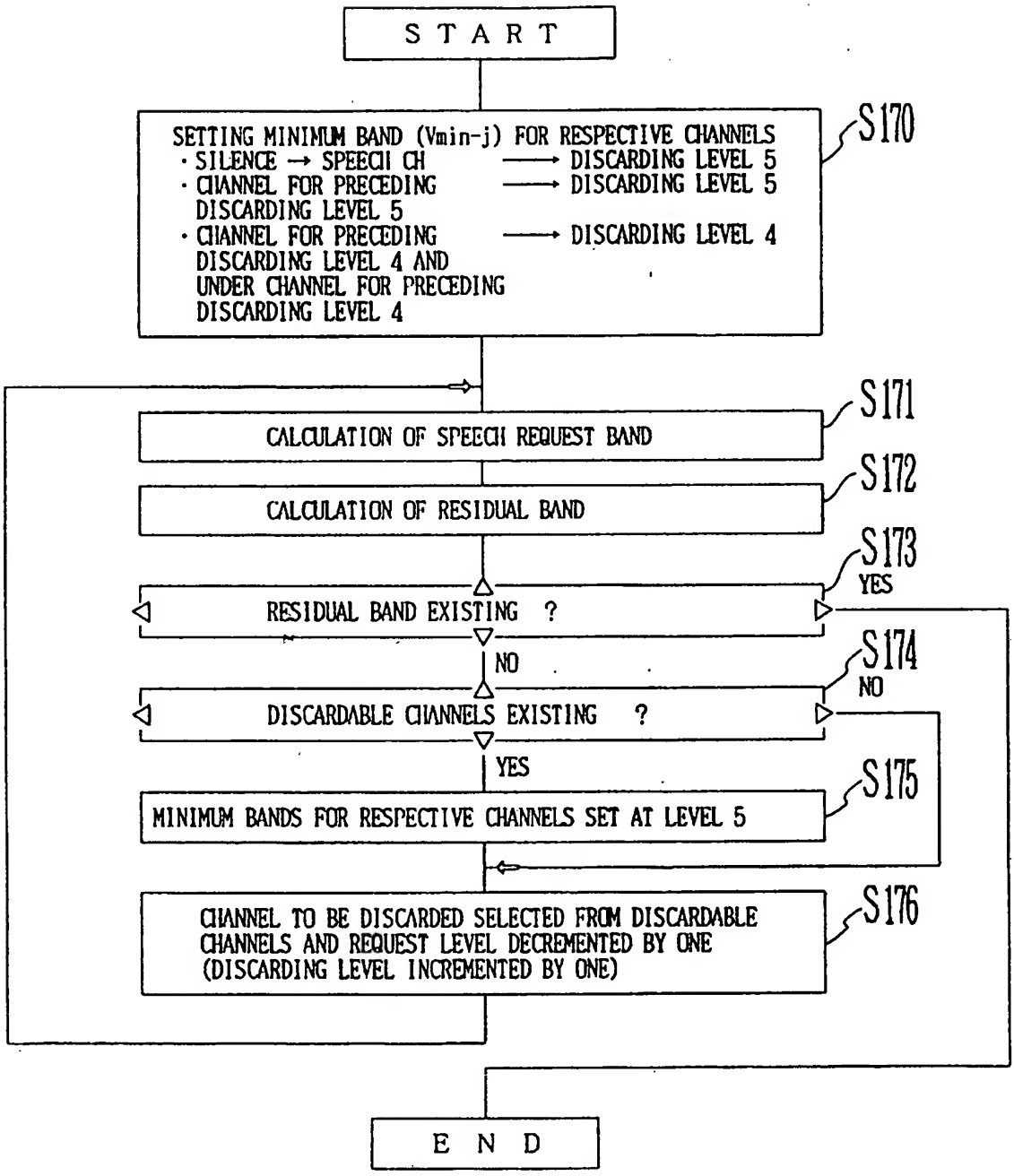
START
SETTING MINIMUM BAND (Vmin-j) FOR RESPECTIVE CHANNELS
· SILENCE → SPEECH CH ⟶ DISCARDING LEVEL 5
· CHANNEL FOR PRECEDING DISCARDING LEVEL 5 ⟶ DISCARDING LEVEL 5
· CHANNEL FOR PRECEDING DISCARDING LEVEL 4 AND UNDER CHANNEL FOR PRECEDING DISCARDING LEVEL 4 ⟶ DISCARDING LEVEL 4
S170
CALCULATION OF SPEECH REQUEST BAND
S171
CALCULATION OF RESIDUAL BAND
S172
RESIDUAL BAND EXISTING ?
S173
YES
NO
DISCARDABLE CHANNELS EXISTING ?
S174
NO
YES
MINIMUM BANDS FOR RESPECTIVE CHANNELS SET AT LEVEL 5
S175
CHANNEL TO BE DISCARDED SELECTED FROM DISCARDABLE CHANNELS AND REQUEST LEVEL DECREMENTED BY ONE (DISCARDING LEVEL INCREMENTED BY ONE)
S176
END

Fig. 41

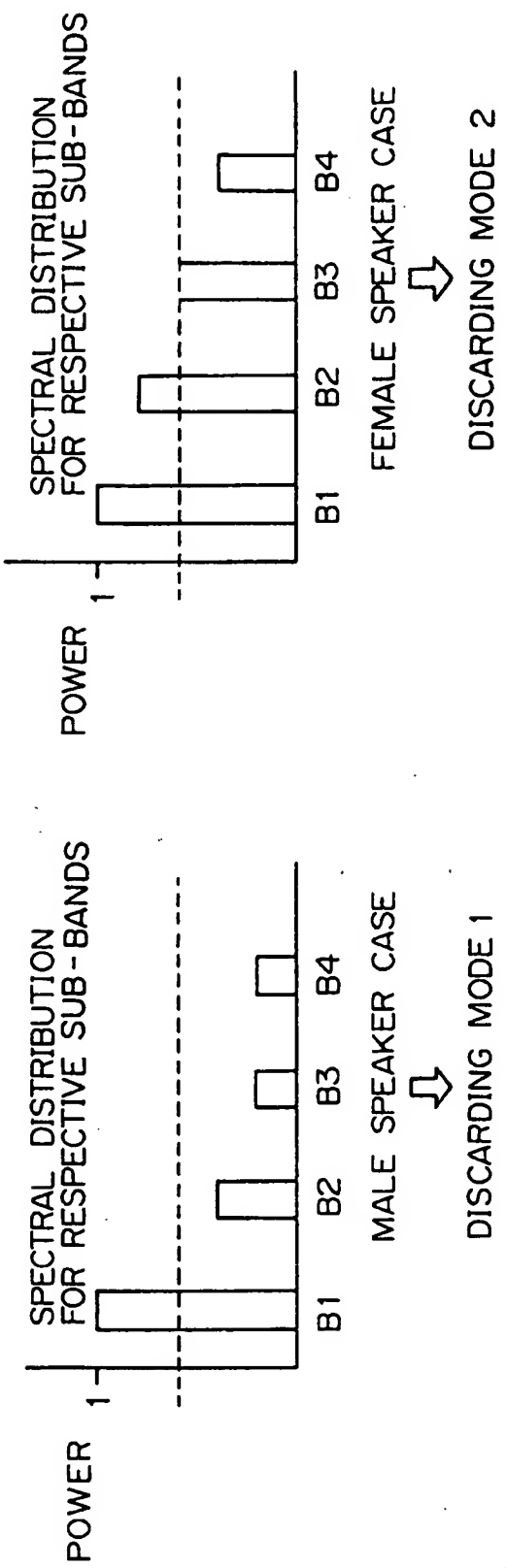

Fig. 42A

Fig. 42B

ITEM	OPERATING MODE		MODE 1(MALE) (S,B1,B2,B3,B4)	MODE 2(FEMALE) (S,B1,B2,B3,B4)		BETWEEN MUX AND DMUX RI2~0
(1)	NOT IN USE		(0,0,0,0,0)	(0,0,0,0,0)	0kbps	000
(2)	SILENCE TIME		(0,0,0,0,0)	(0,0,0,0,0)	0kbps	001
(3)	SPEECH TIME	DISCARDING LEVEL 5	(1,2,2,2,0)	(1,2,2,2,0)	13kbps	010
(4)		DISCARDING LEVEL 4	(1,4,2,2,0)	(1,2,2,2,2)	17kbps	011
(5)		DISCARDING LEVEL 3	(1,4,4,3,2)	(1,4,4,3,2)	27kbps	100
(6)		DISCARDING LEVEL 2	(1,5,5,5,3)	(1,5,5,4,4)	37kbps	101
(7)		DISCARDING LEVEL 1	(1,5,5,5,5)	(1,5,5,5,5)	41kbps	110

SIDE INFORMATION IS USED FOR IDENTIFYING MODES AND 1kbps IS USED

Fig. 43

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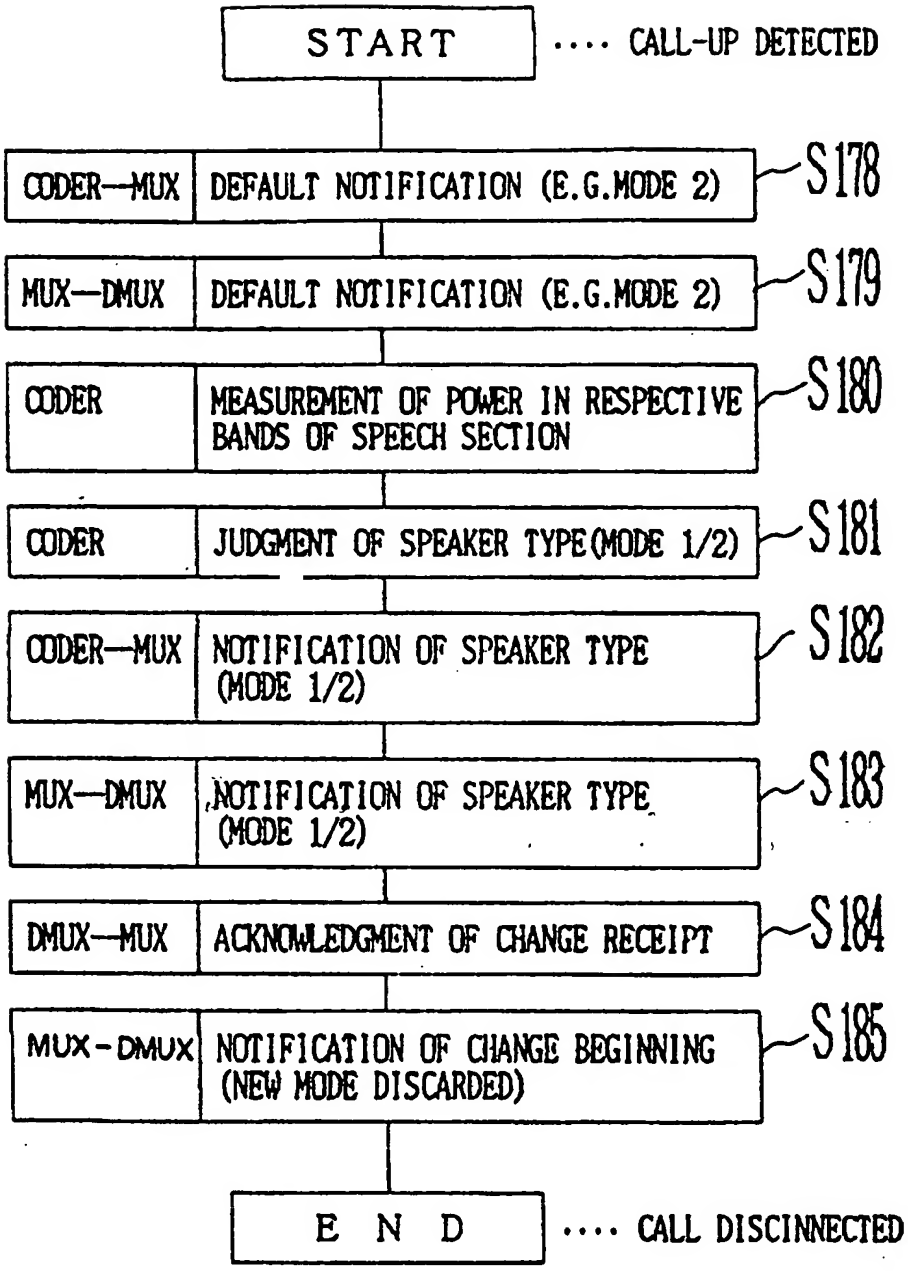
START
···· CALL-UP DETECTED
CODER—MUX
DEFAULT NOTIFICATION (E.G.MODE 2)
S178
MUX—DMUX
DEFAULT NOTIFICATION (E.G.MODE 2)
S179
CODER
MEASUREMENT OF POWER IN RESPECTIVE BANDS OF SPEECH SECTION
S180
CODER
JUDGMENT OF SPEAKER TYPE(MODE 1/2)
S181
CODER—MUX
NOTIFICATION OF SPEAKER TYPE (MODE 1/2)
S182
MUX—DMUX
NOTIFICATION OF SPEAKER TYPE (MODE 1/2)
S183
DMUX—MUX
ACKNOWLEDGMENT OF CHANGE RECEIPT
S184
MUX-DMUX
NOTIFICATION OF CHANGE BEGINNING (NEW MODE DISCARDED)
S185
E N D
···· CALL DISCINNECTED

Fig. 44A

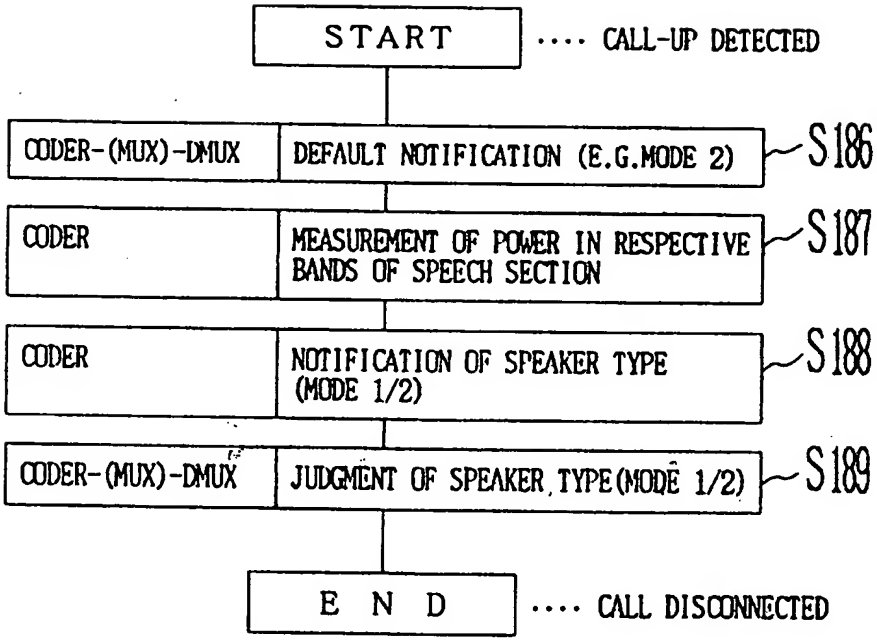
START
···· CALL-UP DETECTED
CODER-(MUX)-DMUX
DEFAULT NOTIFICATION (E.G.MODE 2)
S186
CODER
MEASUREMENT OF POWER IN RESPECTIVE BANDS OF SPEECH SECTION
S187
CODER
NOTIFICATION OF SPEAKER TYPE (MODE 1/2)
S188
CODER-(MUX)-DMUX
JUDGMENT OF SPEAKER TYPE(MODE 1/2)
S189
E N D
···· CALL DISCONNECTED

Fig. 44B

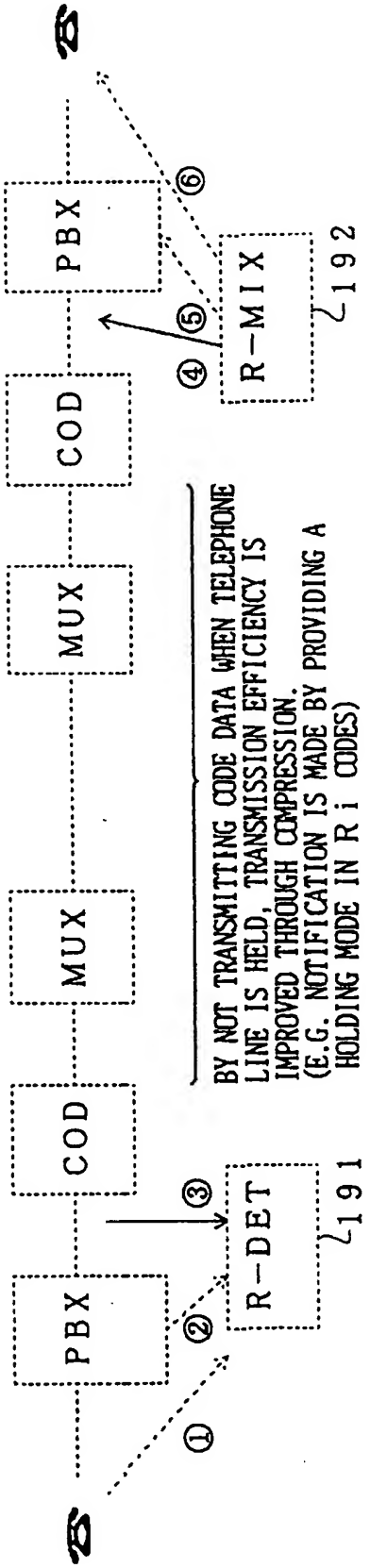
PBX
COD
MUX
MUX
COD
PBX
R-DET
R-MIX
191
192
①
②
③
④
⑤
⑥
BY NOT TRANSMITTING CODE DATA WHEN TELEPHONE LINE IS HELD, TRANSMISSION EFFICIENCY IS IMPROVED THROUGH COMPRESSION. (E.G. NOTIFICATION IS MADE BY PROVIDING A HOLDING MODE IN Ri CODES)

Fig. 45

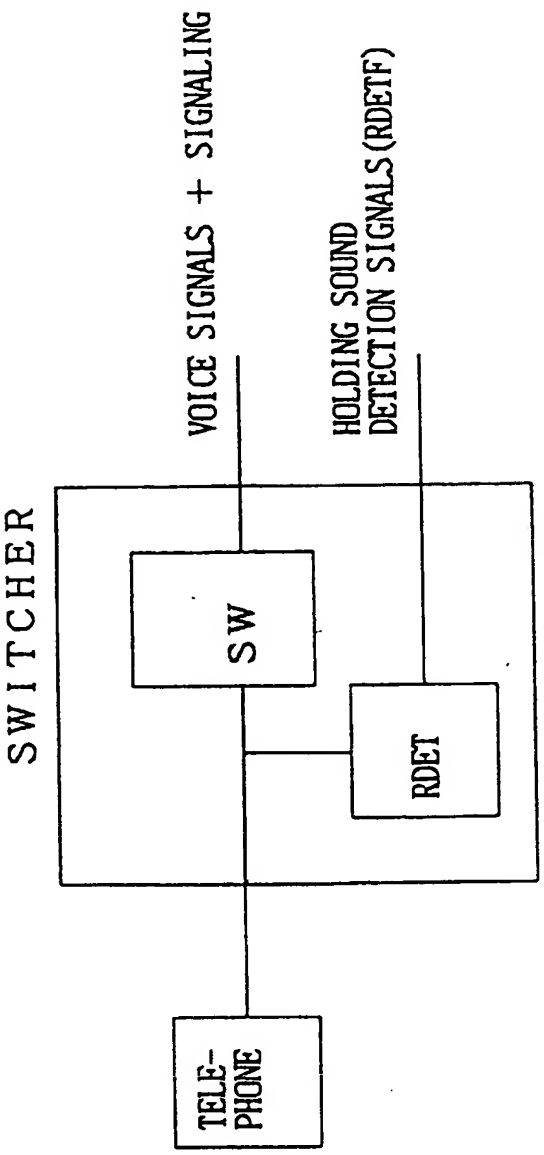
SWITCHER
SW
RDET
VOICE SIGNALS + SIGNALING
HOLDING SOUND
DETECTION SIGNALS(RDETF)
TELE-
PHONE

Fig. 46A

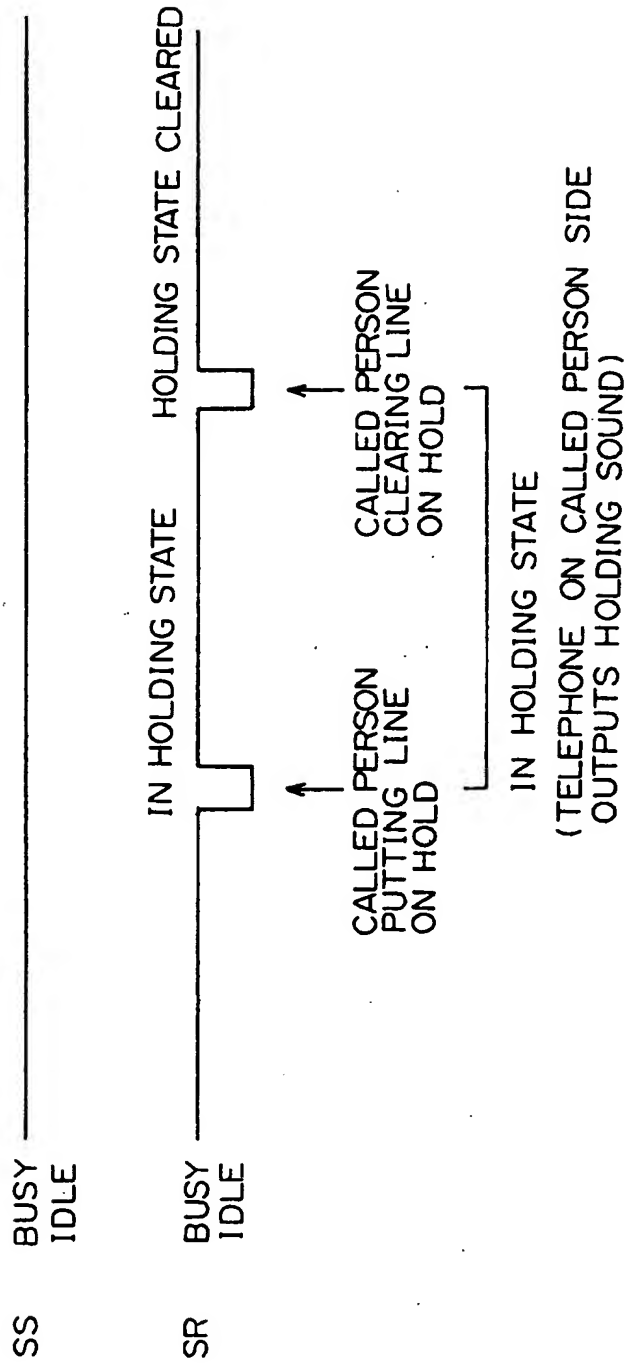
SS
BUSY
IDLE
SR
BUSY
IDLE
IN HOLDING STATE
HOLDING STATE CLEARED
CALLED PERSON PUTTING LINE ON HOLD
CALLED PERSON CLEARING LINE ON HOLD
IN HOLDING STATE
(TELEPHONE ON CALLED PERSON SIDE OUTPUTS HOLDING SOUND)

Fig. 46B

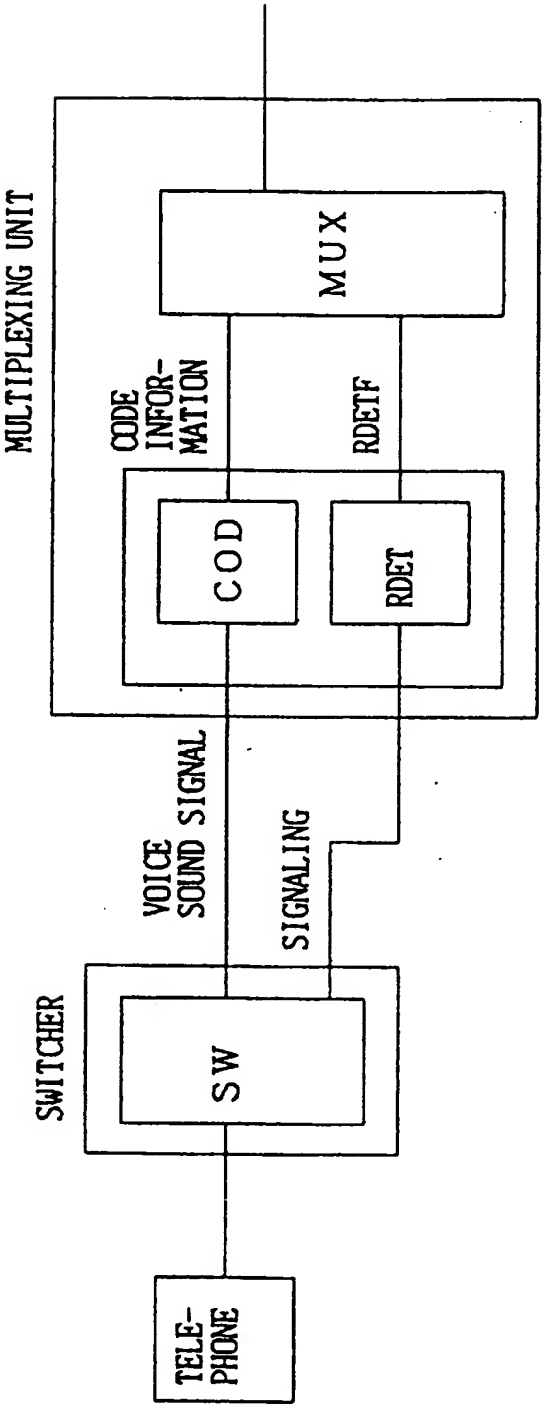
MULTIPLEXING UNIT
MUX
CODE INFOR-MATION
RDETF
COD
RDET
VOICE SOUND SIGNAL
SIGNALING
SWITCHER
SW
TELE-PHONE

Fig. 47

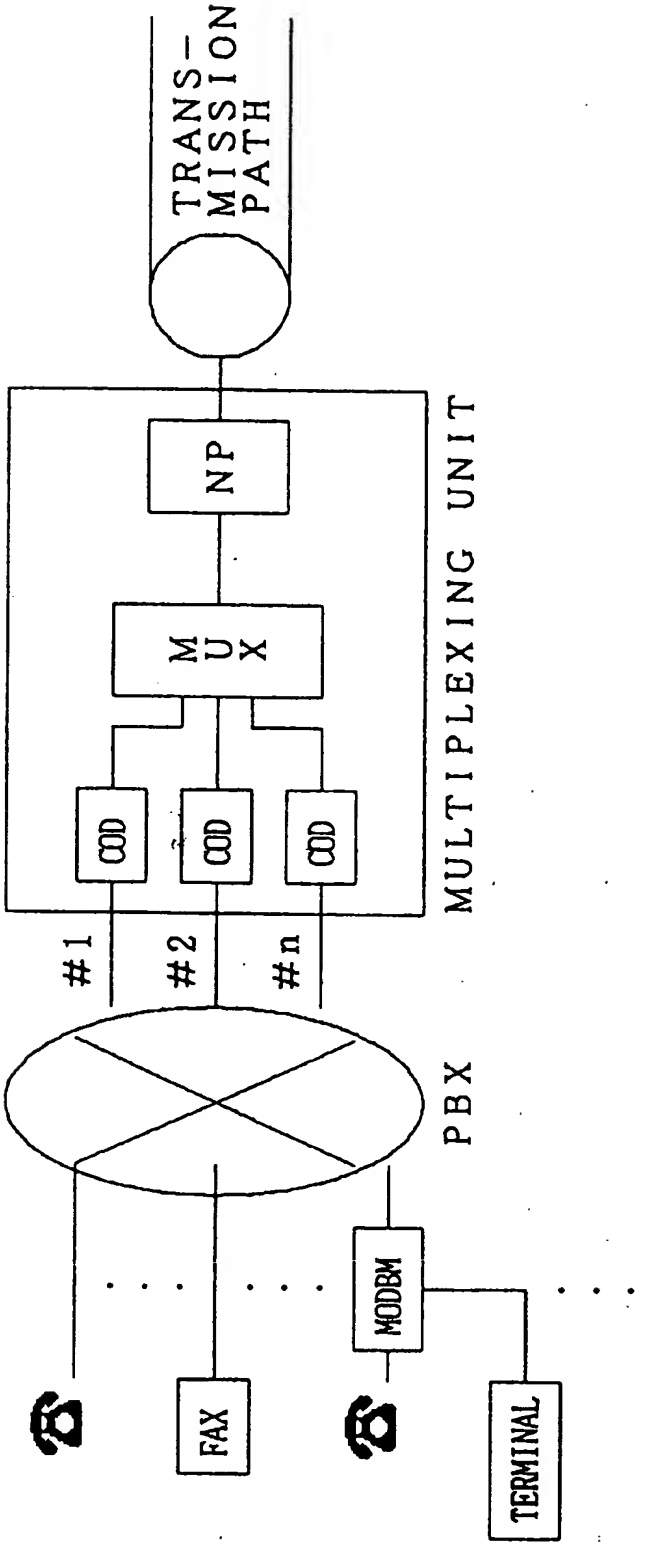
TRANS-
MISSION
PATH
NP
MUX
CODD
CODD
CODD
MULTIPLEXING UNIT
#1
#2
#n
PBX
MODEM
FAX
TERMINAL

Fig. 48

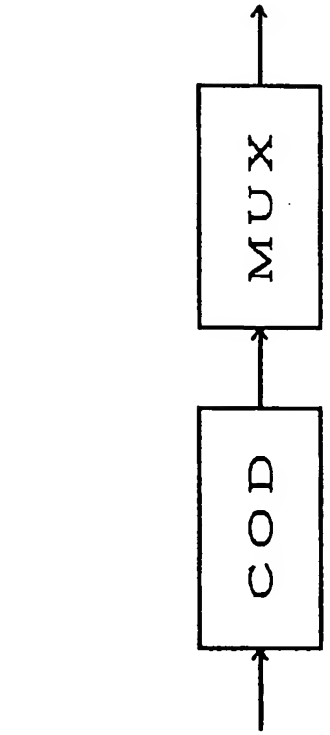

Fig. 49B

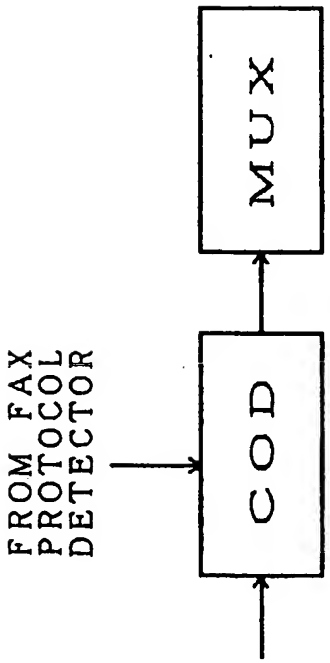

Fig. 49A

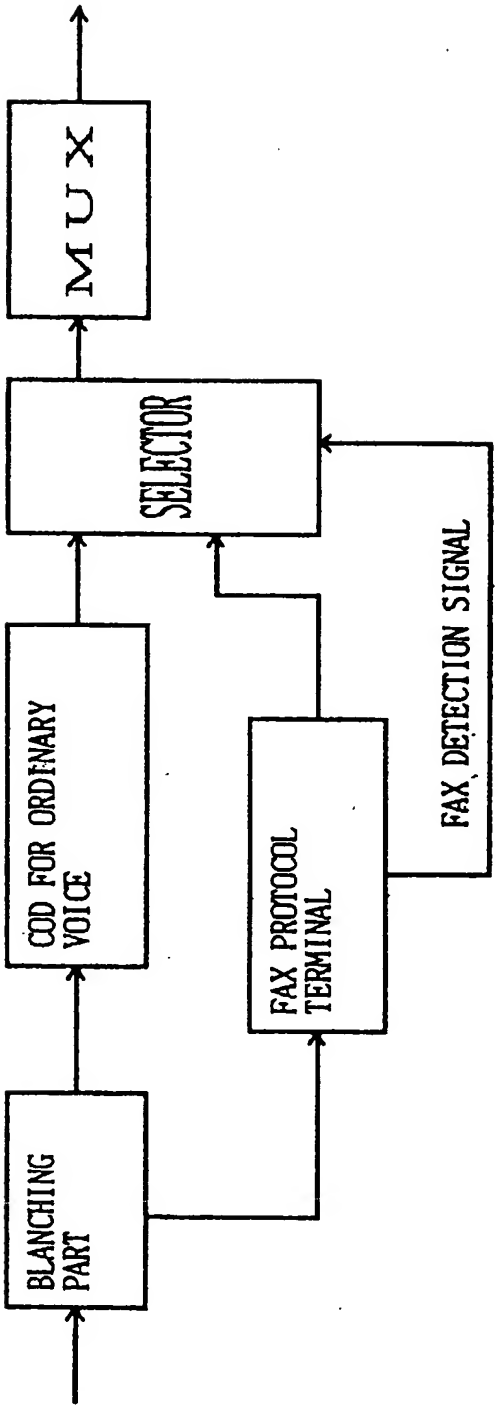
MUX
SELECTOR
COD FOR ORDINARY VOICE
FAX PROTOCOL TERMINAL
FAX DETECTION SIGNAL
BLANCHING PART

Fig. 49C

[PHASE A]
  CALLING SIDE SENDING CNG 1100 Hz (FAX IDENTIFICATION SIGNALS)
  CALLED SIDE   SENDING CED 2100 Hz (RESPONSE SIGNALS)
[PHASE B]
  CALLED SIDE   RECEIVING SIDE EQUIPPED FUNCTION NOTIFIED
  CALLING SIDE FUNCTIONS TO BE USED NOTIFIED
  CALLING SIDE TRAINING SIGNALS
  CALLED SIDE   TRAINING RESULT NOTIFIED (OK/NG)
[PHASE C]
  CALLING SIDE TRANSMISSION OF IMAGE INFORMATION FOR ONE PAGE
[PHASE D]
  CALLING SIDE NOTIFYING COMPLETE TRANSMISSION OF ONE PAGE
  CALLED SIDE   RESPONDING RESULT OF RECEIPT
THEREAFTER [PHASE C AND D] REPEATED FOR NUMBER OF PAGES
[PHASE E]
  CALLING SIDE DECLARING TERMINATION

Fig. 50

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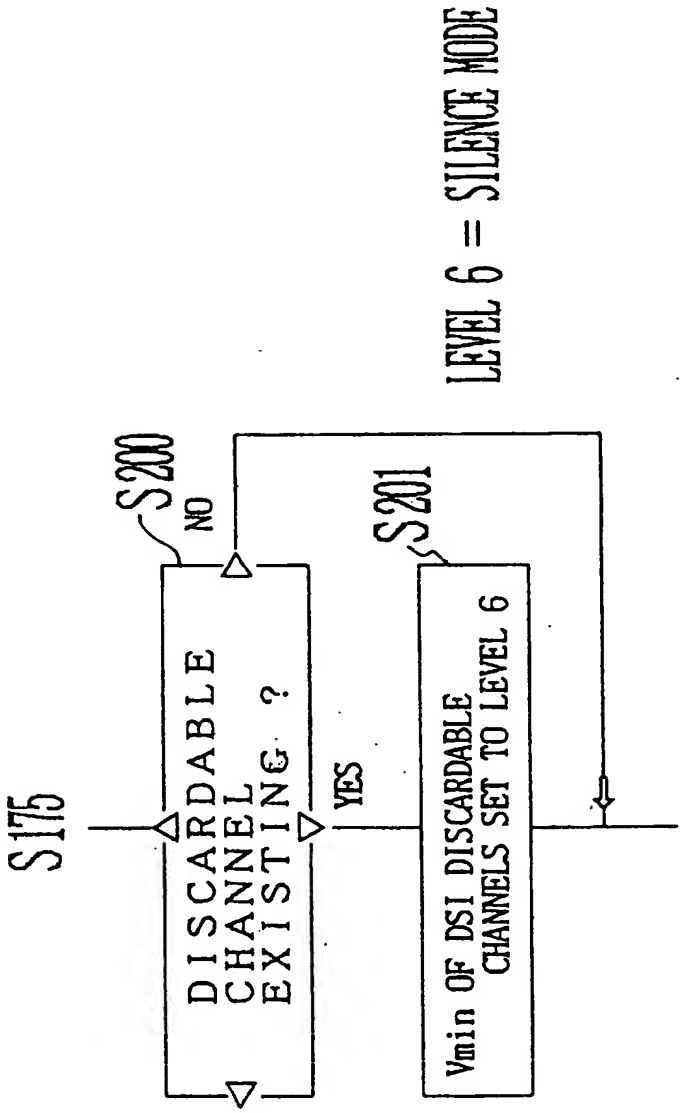
LEVEL 6 = SILENCE MODE
S175
S200
DISCARDABLE CHANNEL EXISTING ?
NO
YES
S201
Vmin OF DSI DISCARDABLE CHANNELS SET TO LEVEL 6
S176

Fig. 51

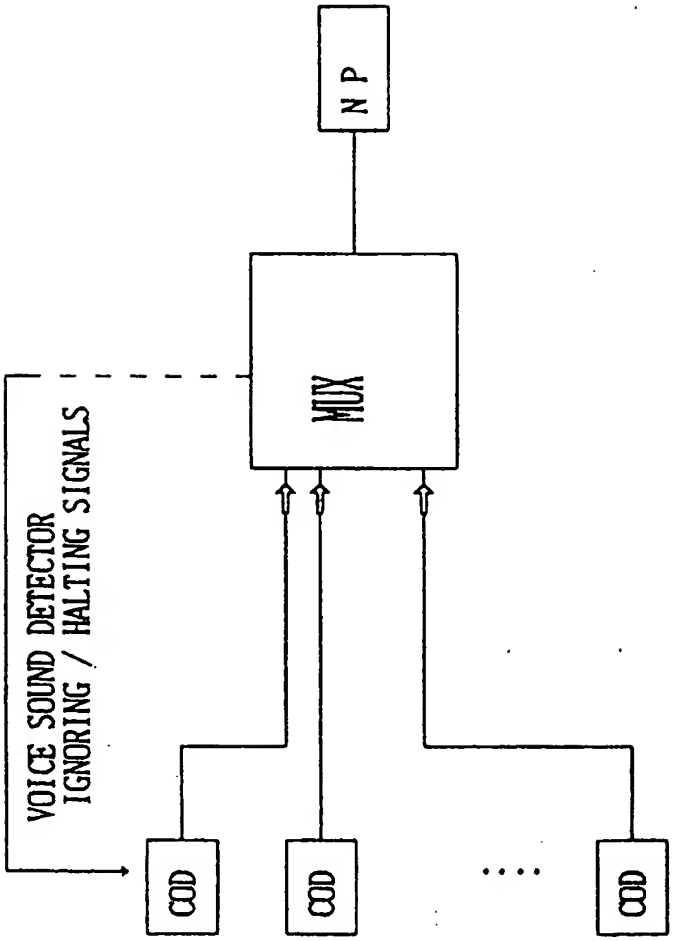
N P
MUX
VOICE SOUND DETECTOR
IGNORING / HALTING SIGNALS
COD
COD
COD

Fig. 52

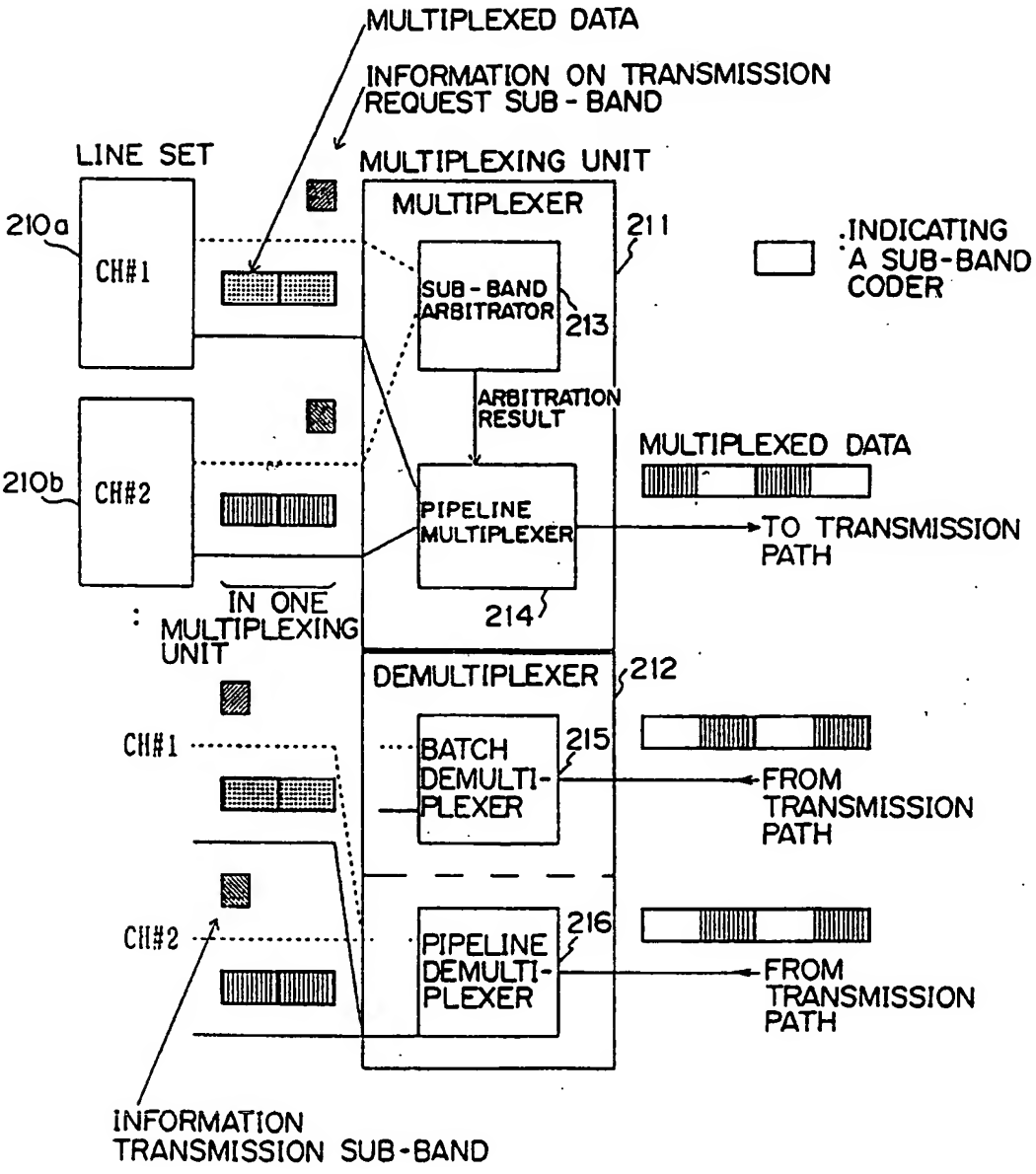
MULTIPLEXED DATA
INFORMATION ON TRANSMISSION REQUEST SUB-BAND
LINE SET
MULTIPLEXING UNIT
MULTIPLEXER
211
210a
CH#1
SUB-BAND ARBITRATOR
213
INDICATING A SUB-BAND CODER
ARBITRATION RESULT
MULTIPLEXED DATA
210b
CH#2
PIPELINE MULTIPLEXER
TO TRANSMISSION PATH
IN ONE MULTIPLEXING UNIT
214
DEMULTIPLEXER
212
CH#1
BATCH DEMULTI-PLEXER
215
FROM TRANSMISSION PATH
CH#2
PIPELINE DEMULTI-PLEXER
216
FROM TRANSMISSION PATH
INFORMATION TRANSMISSION SUB-BAND

Fig. 53

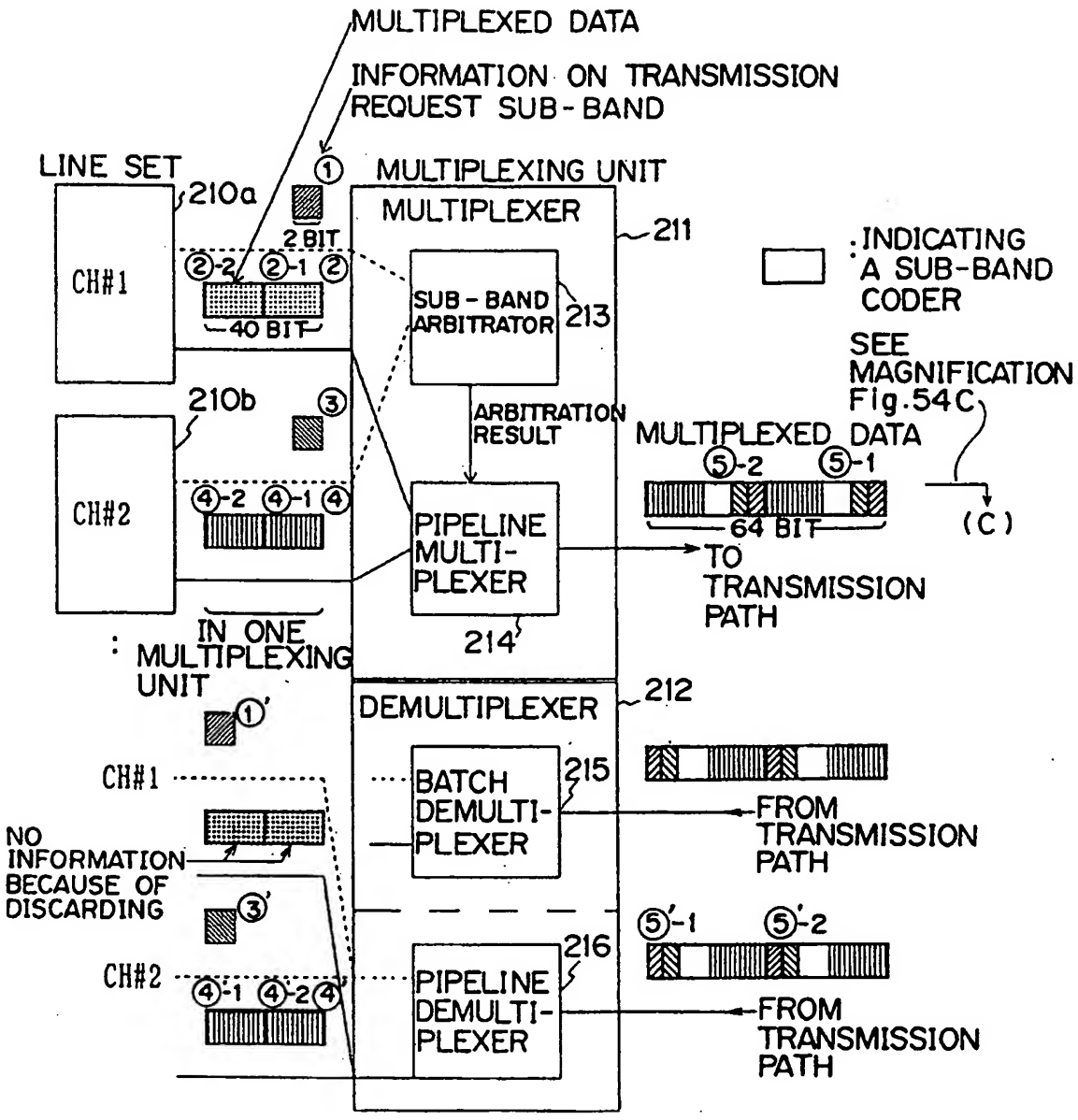
MULTIPLEXED DATA
INFORMATION ON TRANSMISSION REQUEST SUB-BAND
LINE SET
MULTIPLEXING UNIT
210a
MULTIPLEXER
211
2 BIT
CH#1
40 BIT
SUB-BAND ARBITRATOR
213
:INDICATING A SUB-BAND CODER
SEE MAGNIFICATION Fig.54C
210b
ARBITRATION RESULT
MULTIPLEXED DATA
64 BIT
CH#2
PIPELINE MULTI-PLEXER
TO TRANSMISSION PATH
(C)
214
IN ONE MULTIPLEXING UNIT
DEMULTIPLEXER
212
CH#1
BATCH DEMULTI-PLEXER
215
FROM TRANSMISSION PATH
NO INFORMATION BECAUSE OF DISCARDING
CH#2
216
PIPELINE DEMULTI-PLEXER
FROM TRANSMISSION PATH

Fig. 54A

	SUB-BAND ALLOCATED FOR ONE MULTIPLEXING UNIT	SUB-BAND ALLOCATED TO EQUIVALENCE OF ONE SPLIT	EXAMPLES OF INFORMATION ON EXISTENCE OF DATA TRANSMISSION
CH#1	0 BIT	0 BIT	10→00(BECAUSE OF DISCARDING)
CH#2	40 BIT	20 BIT	10→10(AS IS)

Fig. 54B

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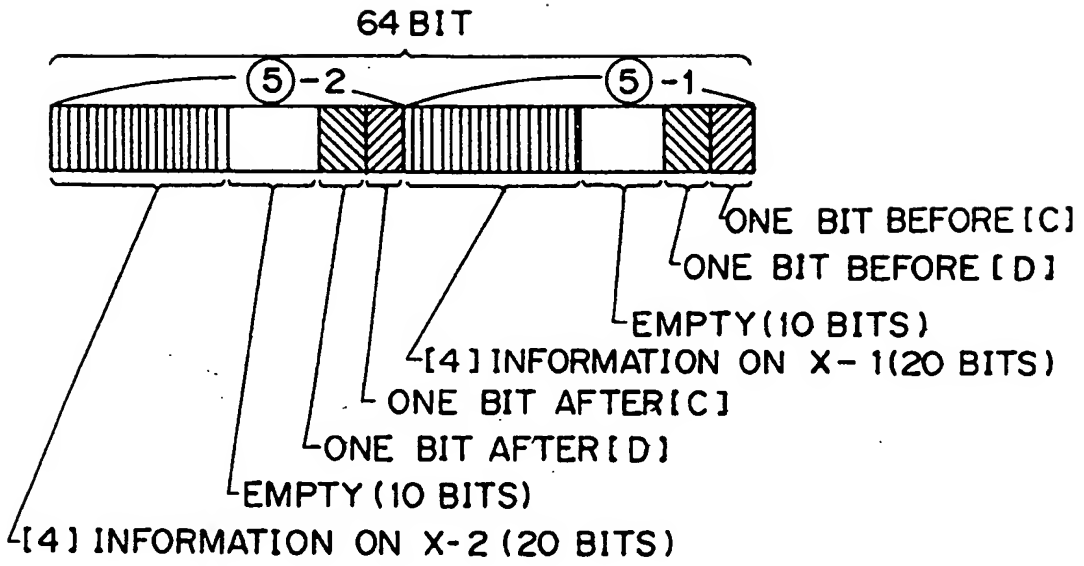

64 BIT
⑤-2
⑤-1
ONE BIT BEFORE [C]
ONE BIT BEFORE [D]
EMPTY (10 BITS)
[4] INFORMATION ON X-1 (20 BITS)
ONE BIT AFTER [C]
ONE BIT AFTER [D]
EMPTY (10 BITS)
[4] INFORMATION ON X-2 (20 BITS)

Fig. 54C

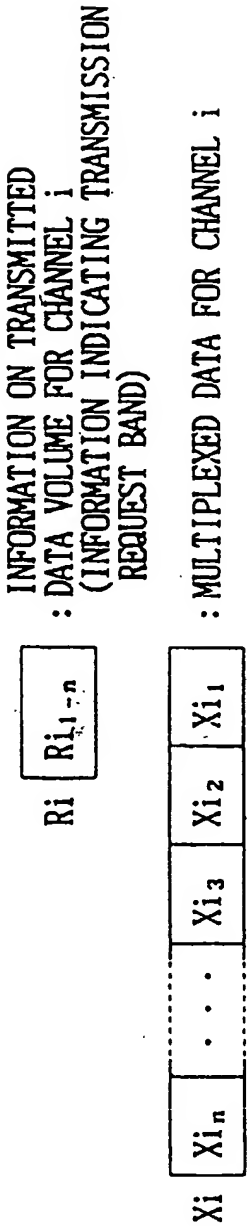

Ri
Ri1-n
: INFORMATION ON TRANSMITTED DATA VOLUME FOR CHANNEL i (INFORMATION INDICATING TRANSMISSION REQUEST BAND)
Xi
Xin
. . .
Xi3
Xi2
Xi1
: MULTIPLEXED DATA FOR CHANNEL i

Fig. 55A

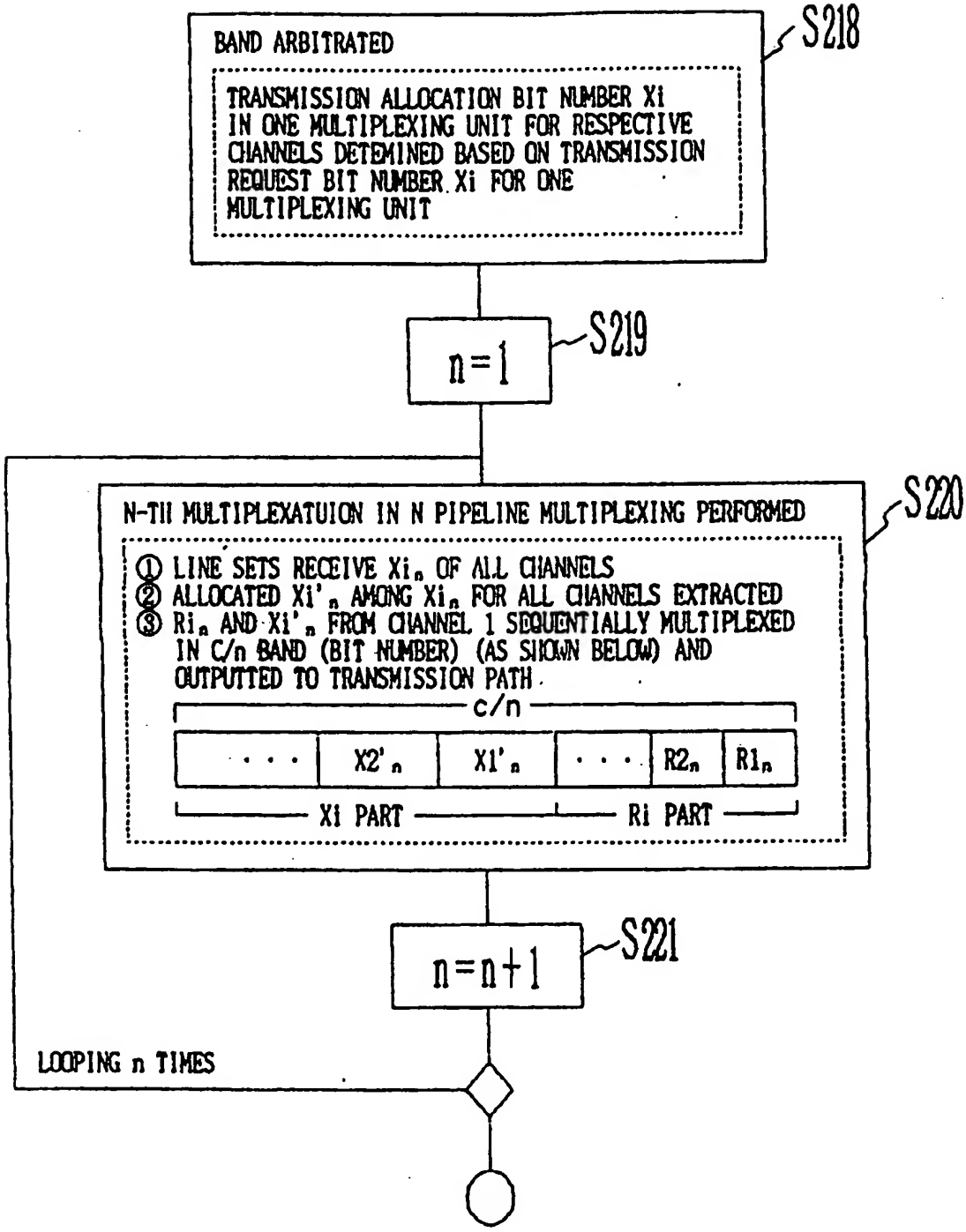

BAND ARBITRATED
TRANSMISSION ALLOCATION BIT NUMBER Xi IN ONE MULTIPLEXING UNIT FOR RESPECTIVE CHANNELS DETEMINED BASED ON TRANSMISSION REQUEST BIT NUMBER Xi FOR ONE MULTIPLEXING UNIT
S218
n=1
S219
N-TH MULTIPLEXATUION IN N PIPELINE MULTIPLEXING PERFORMED
① LINE SETS RECEIVE Xi n OF ALL CHANNELS
② ALLOCATED Xi' n AMONG Xi n FOR ALL CHANNELS EXTRACTED
③ Ri n AND Xi' n FROM CHANNEL 1 SEQUENTIALLY MULTIPLEXED IN C/n BAND (BIT NUMBER) (AS SHOWN BELOW) AND OUTPUTTED TO TRANSMISSION PATH
c/n
· · ·
X2' n
X1' n
· · ·
R2n
R1n
Xi PART
Ri PART
S220
n=n+1
S221
LOOPING n TIMES

Fig. 55B

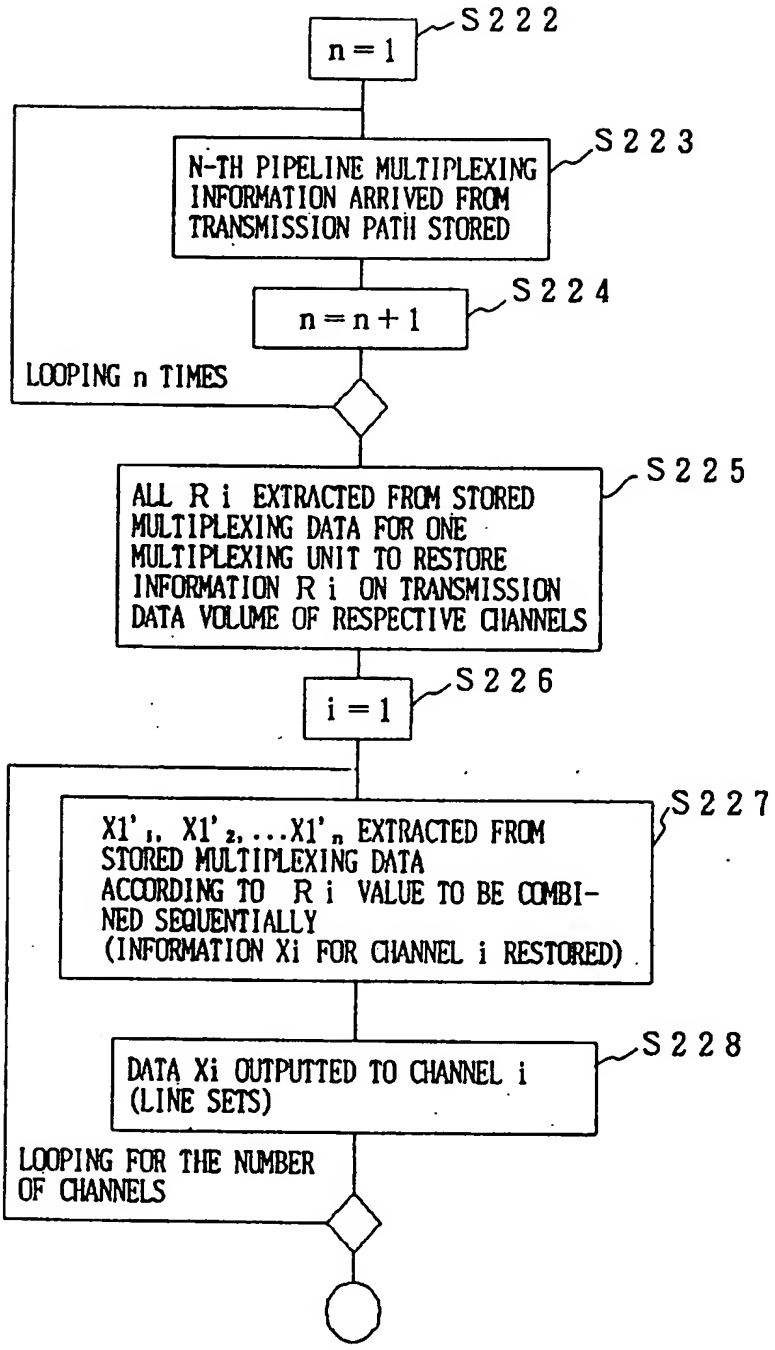

n = 1
S222
N-TH PIPELINE MULTIPLEXING INFORMATION ARRIVED FROM TRANSMISSION PATH STORED
S223
n = n + 1
S224
LOOPING n TIMES
ALL R i EXTRACTED FROM STORED MULTIPLEXING DATA FOR ONE MULTIPLEXING UNIT TO RESTORE INFORMATION R i ON TRANSMISSION DATA VOLUME OF RESPECTIVE CHANNELS
S225
i = 1
S226
X1'1, X1'2, ...X1'n EXTRACTED FROM STORED MULTIPLEXING DATA ACCORDING TO R i VALUE TO BE COMBINED SEQUENTIALLY (INFORMATION Xi FOR CHANNEL i RESTORED)
S227
DATA Xi OUTPUTTED TO CHANNEL i (LINE SETS)
S228
LOOPING FOR THE NUMBER OF CHANNELS

Fig. 56

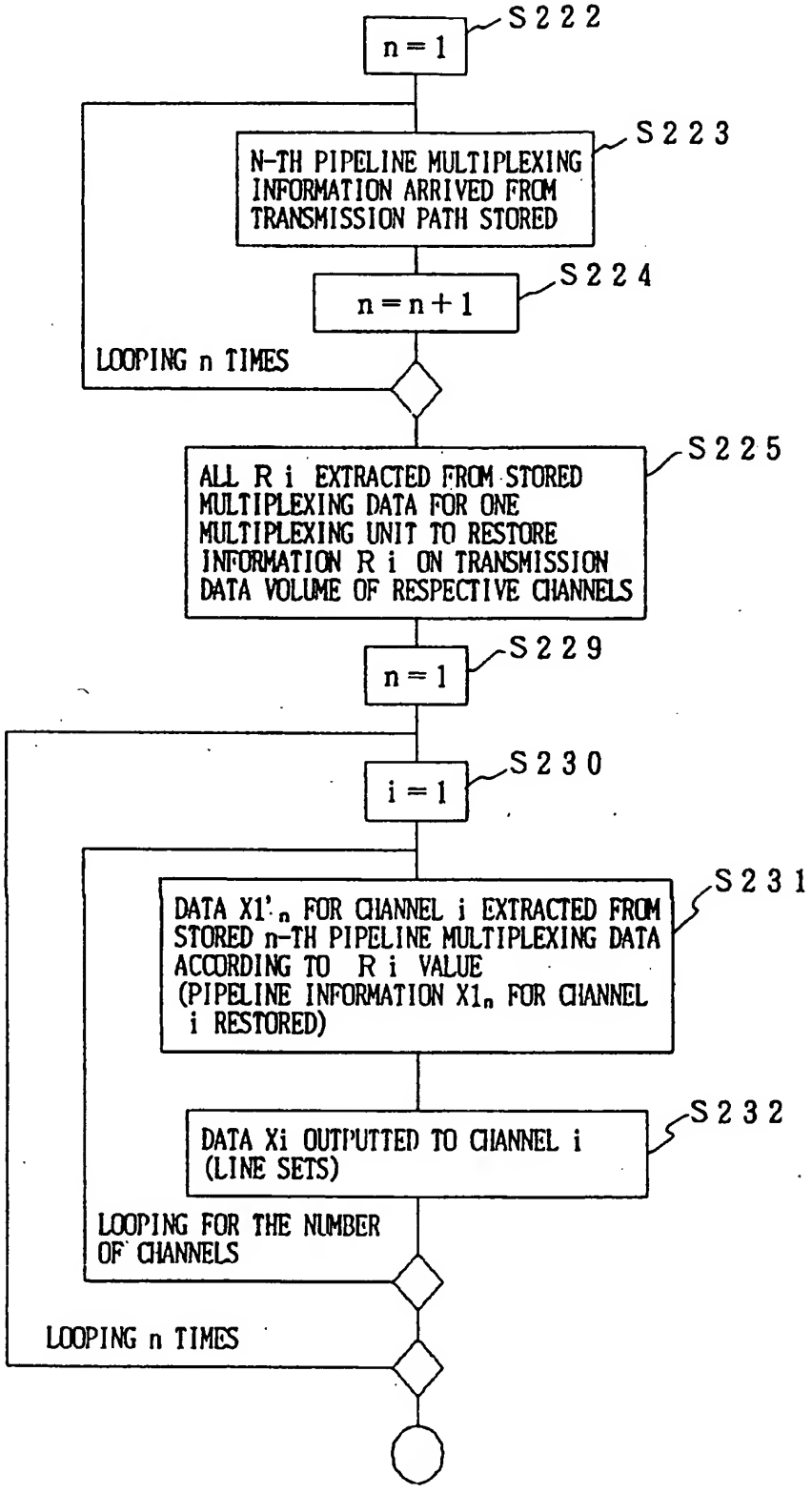

S222
n=1
S223
N-TH PIPELINE MULTIPLEXING INFORMATION ARRIVED FROM TRANSMISSION PATH STORED
S224
n=n+1
LOOPING n TIMES
S225
ALL R i EXTRACTED FROM STORED MULTIPLEXING DATA FOR ONE MULTIPLEXING UNIT TO RESTORE INFORMATION R i ON TRANSMISSION DATA VOLUME OF RESPECTIVE CHANNELS
S229
n=1
S230
i=1
S231
DATA X1'n FOR CHANNEL i EXTRACTED FROM STORED n-TH PIPELINE MULTIPLEXING DATA ACCORDING TO R i VALUE (PIPELINE INFORMATION X1n FOR CHANNEL i RESTORED)
S232
DATA Xi OUTPUTTED TO CHANNEL i (LINE SETS)
LOOPING FOR THE NUMBER OF CHANNELS
LOOPING n TIMES

Fig. 57

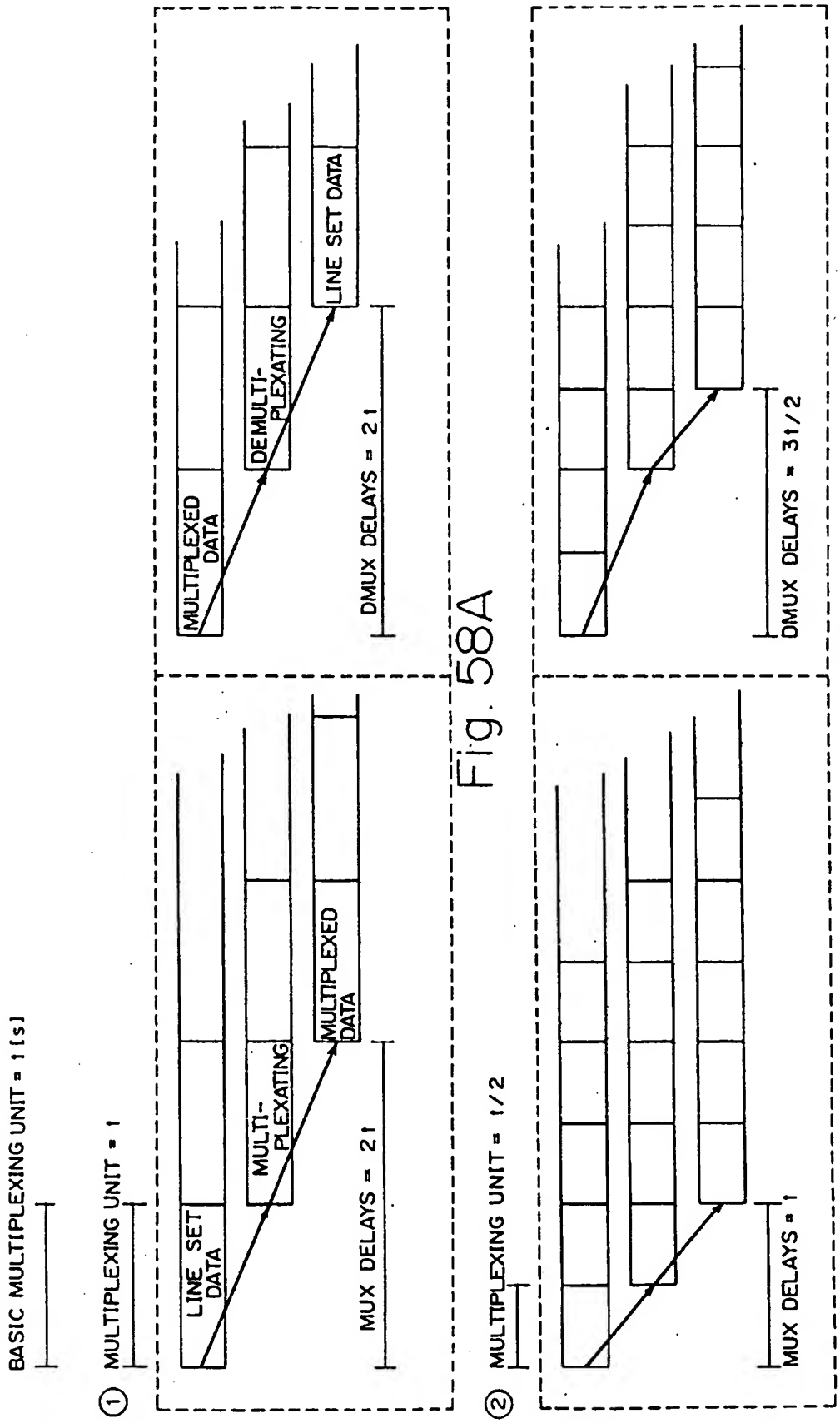


Fig. 58A

Fig. 58B

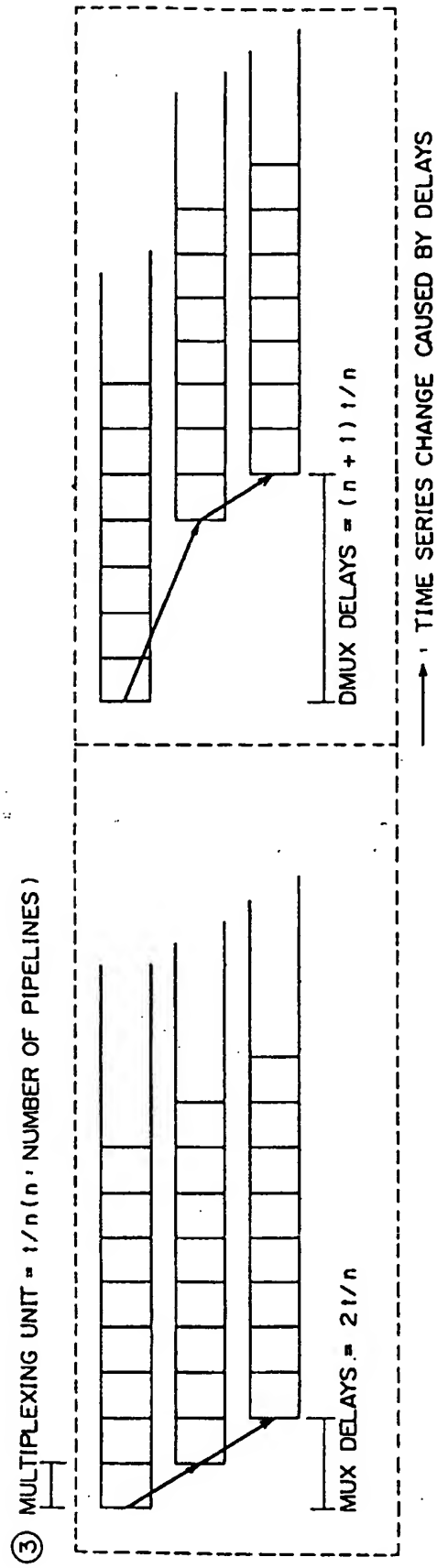
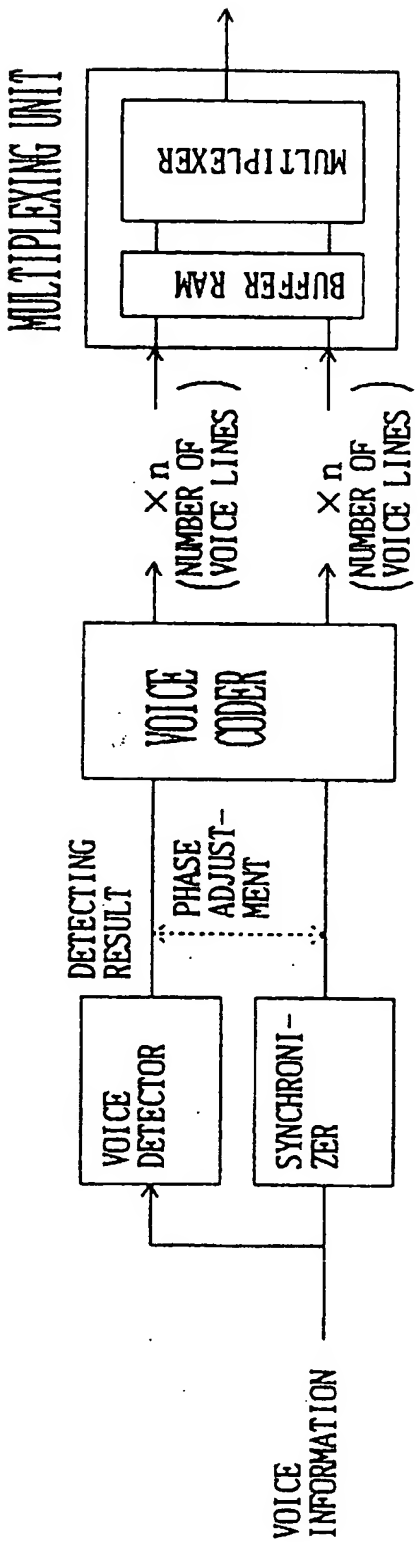
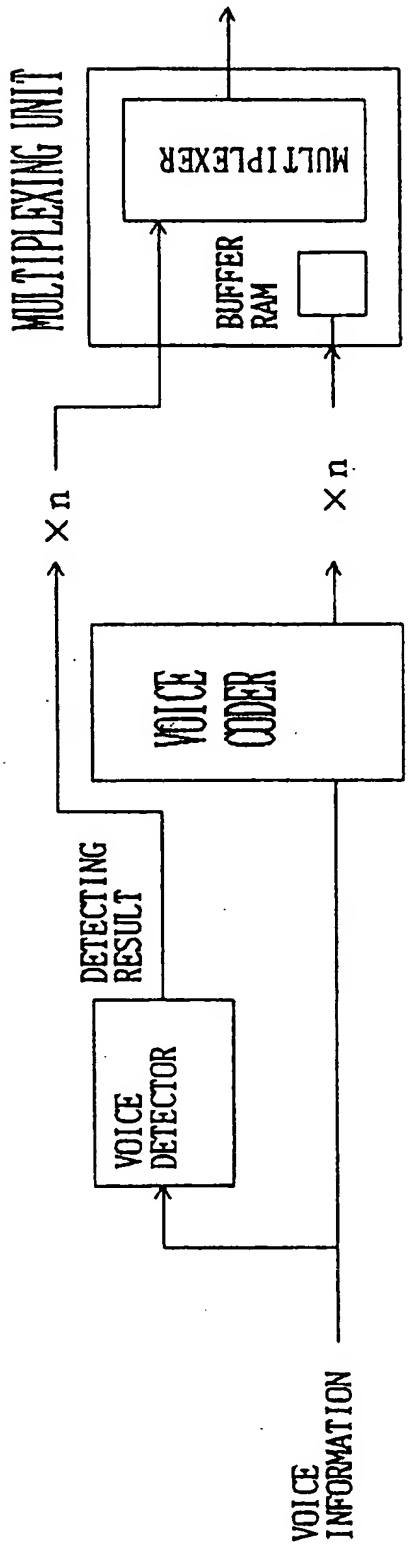

③ MULTIPLEXING UNIT = 1/n (n : NUMBER OF PIPELINES)
MUX DELAYS = 2 1/n
DMUX DELAYS = (n + 1) 1/n
TIME SERIES CHANGE CAUSED BY DELAYS

Fig. 58C



CONVENTIONAL METHOD

Fig. 59A



METHOD OF THIS INVENTION

Fig. 59B

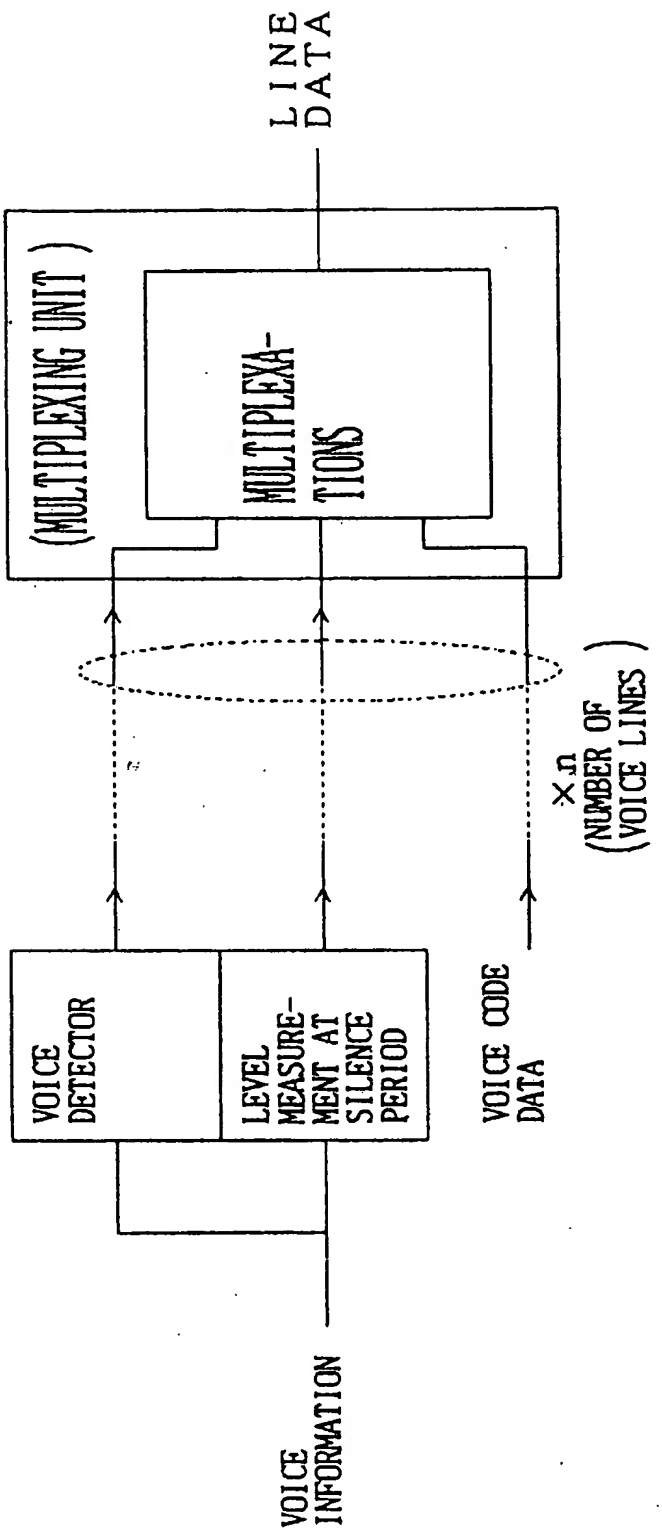

LINE DATA
(MULTIPLEXING UNIT)
MULTIPLEXA-TIONS
×n (NUMBER OF VOICE LINES)
VOICE DETECTOR
LEVEL MEASURE-MENT AT SILENCE PERIOD
VOICE CODE DATA
VOICE INFORMATION

Fig. 60A

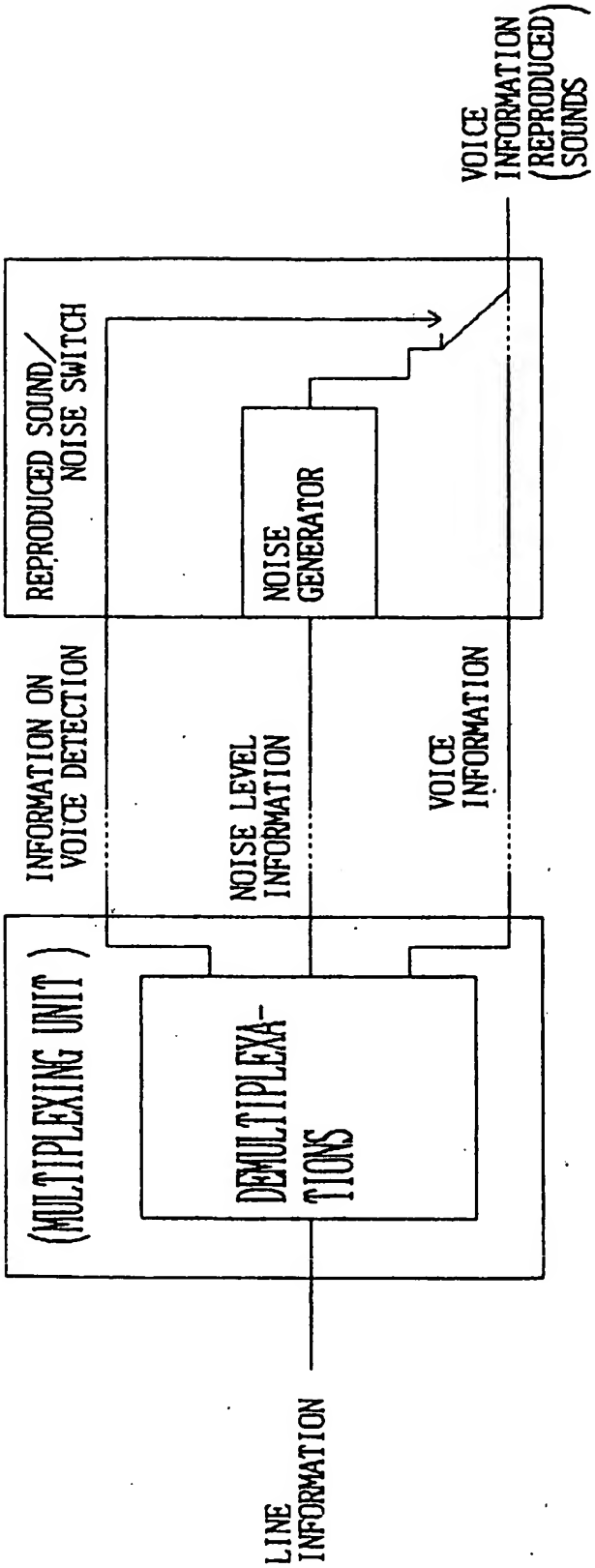

LINE INFORMATION
(MULTIPLEXING UNIT)
DEMULTIPLEXA-TIONS
INFORMATION ON VOICE DETECTION
NOISE LEVEL INFORMATION
VOICE INFORMATION
REPRODUCED SOUND/ NOISE SWITCH
NOISE GENERATOR
VOICE INFORMATION (REPRODUCED SOUNDS)

Fig. 60B

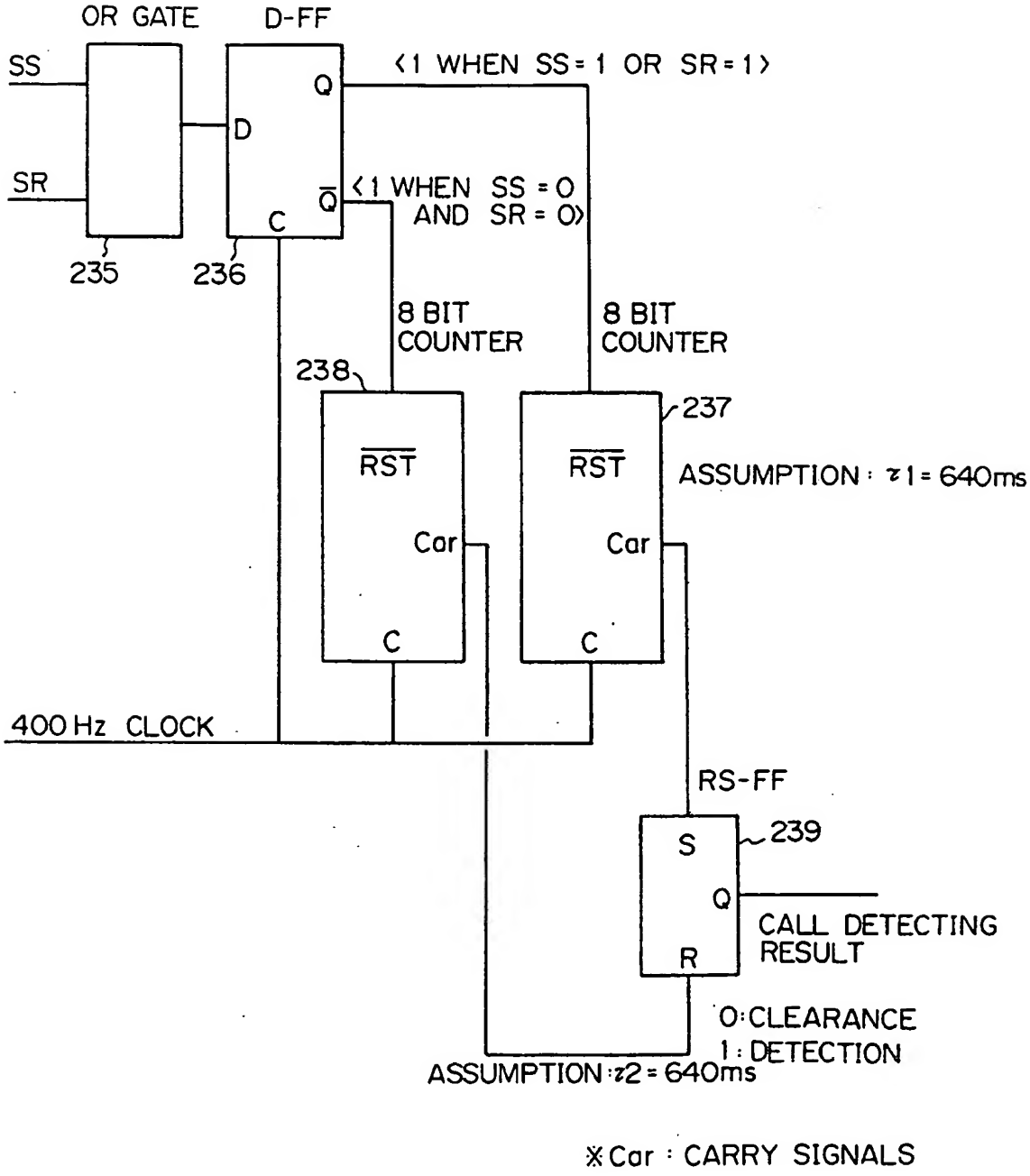

OR GATE
D-FF
SS
SR
D
Q
C
Q̄
〈1 WHEN SS = 1 OR SR = 1〉
〈1 WHEN SS = 0 AND SR = 0〉
235
236
8 BIT COUNTER
8 BIT COUNTER
238
237
RST
RST
Car
Car
C
C
ASSUMPTION : τ1 = 640ms
400 Hz CLOCK
RS-FF
239
S
Q
R
CALL DETECTING RESULT
0:CLEARANCE
1:DETECTION
ASSUMPTION : τ2 = 640ms
※Car : CARRY SIGNALS

Fig. 61

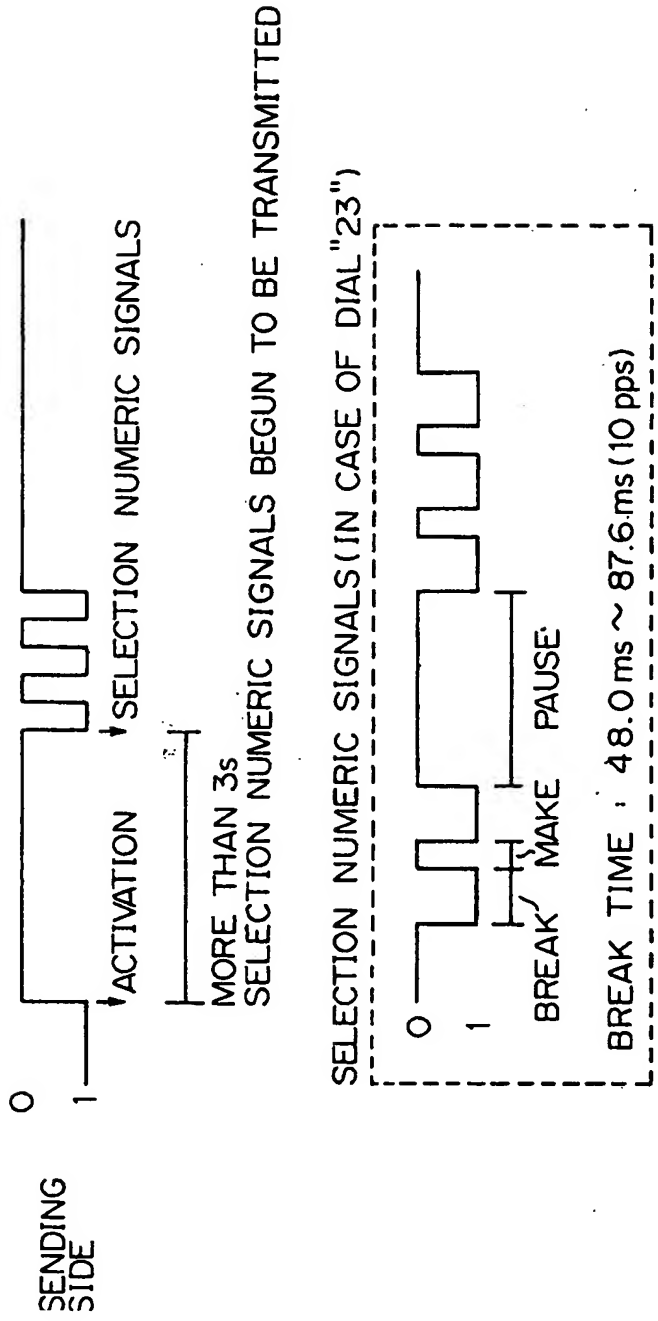

SENDING SIDE
0
1
ACTIVATION
SELECTION NUMERIC SIGNALS
MORE THAN 3s
SELECTION NUMERIC SIGNALS BEGUN TO BE TRANSMITTED
SELECTION NUMERIC SIGNALS (IN CASE OF DIAL "23")
0
1
BREAK
MAKE
PAUSE
BREAK TIME : 48.0 ms ~ 87.6 ms (10 pps)
RESPONDING
RECEIVING SIDE
0
1

Fig. 62

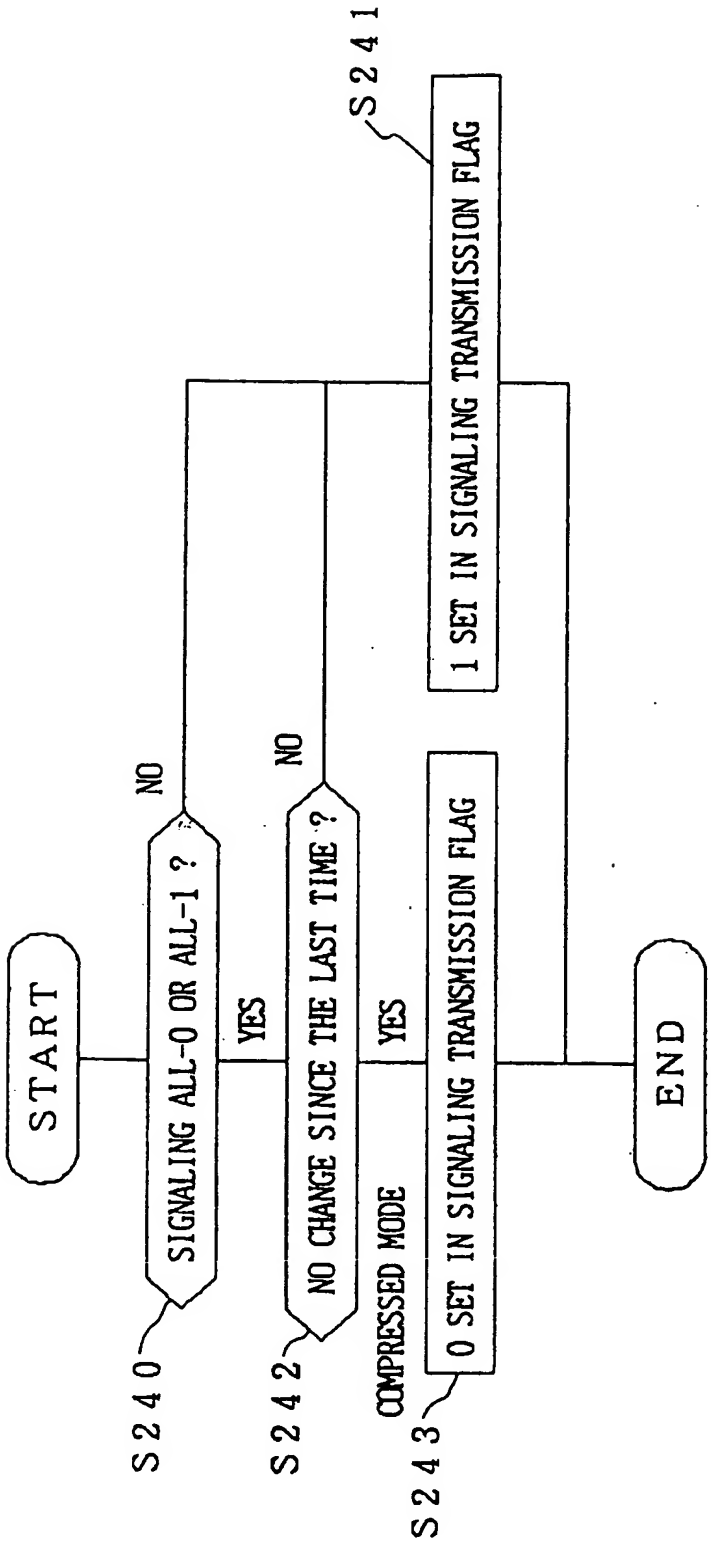

START
SIGNALING ALL-0 OR ALL-1 ?
NO
YES
S240
NO CHANGE SINCE THE LAST TIME ?
NO
YES
S242
COMPRESSED MODE
0 SET IN SIGNALING TRANSMISSION FLAG
S243
1 SET IN SIGNALING TRANSMISSION FLAG
S241
END

Fig. 63A

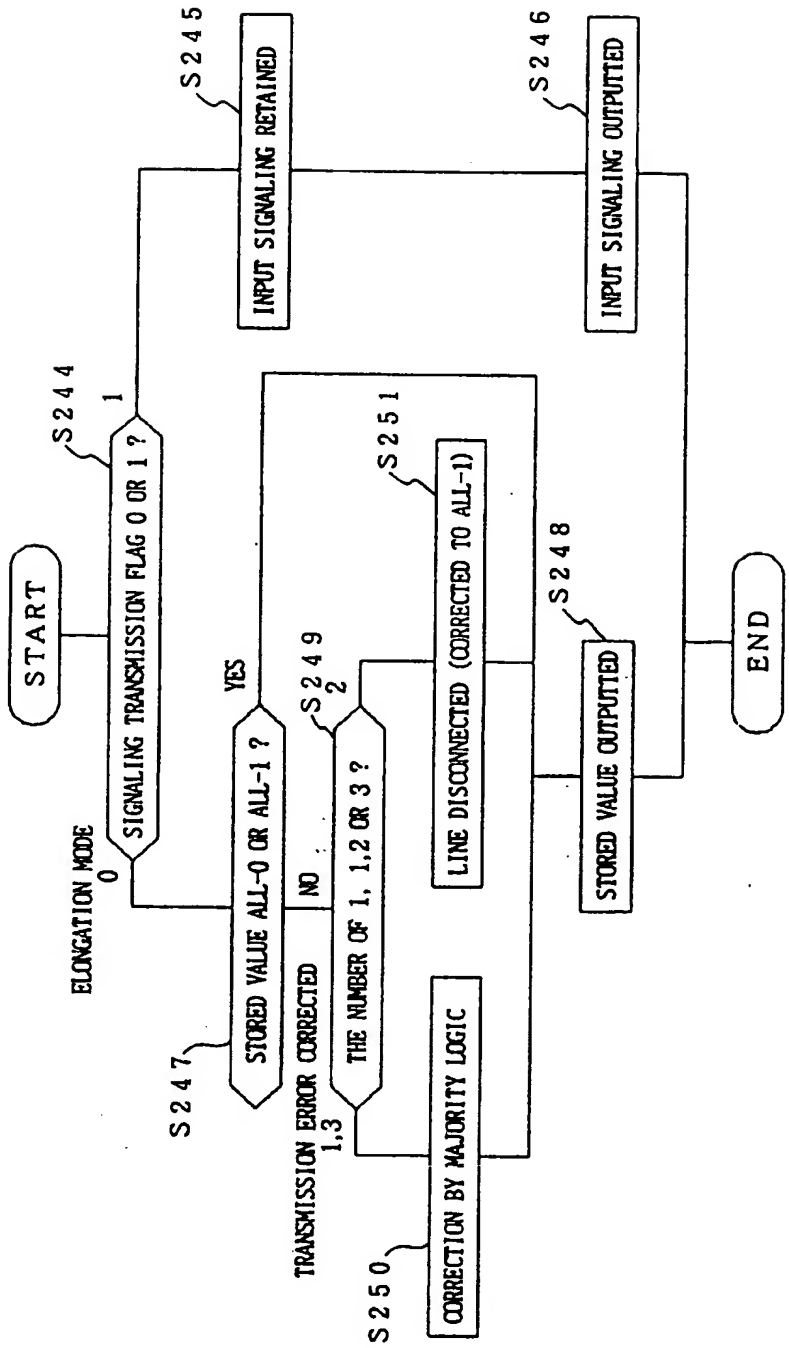

START
S244
SIGNALING TRANSMISSION FLAG 0 OR 1 ?
ELONGATION MODE
0
1
S245
INPUT SIGNALING RETAINED
S246
INPUT SIGNALING OUTPUTTED
S247
STORED VALUE ALL-0 OR ALL-1 ?
YES
NO
TRANSMISSION ERROR CORRECTED
1,3
S249
THE NUMBER OF 1, 1,2 OR 3 ?
2
S251
LINE DISCONNECTED (CORRECTED TO ALL-1)
S248
STORED VALUE OUTPUTTED
END
S250
CORRECTION BY MAJORITY LOGIC

Fig. 63B

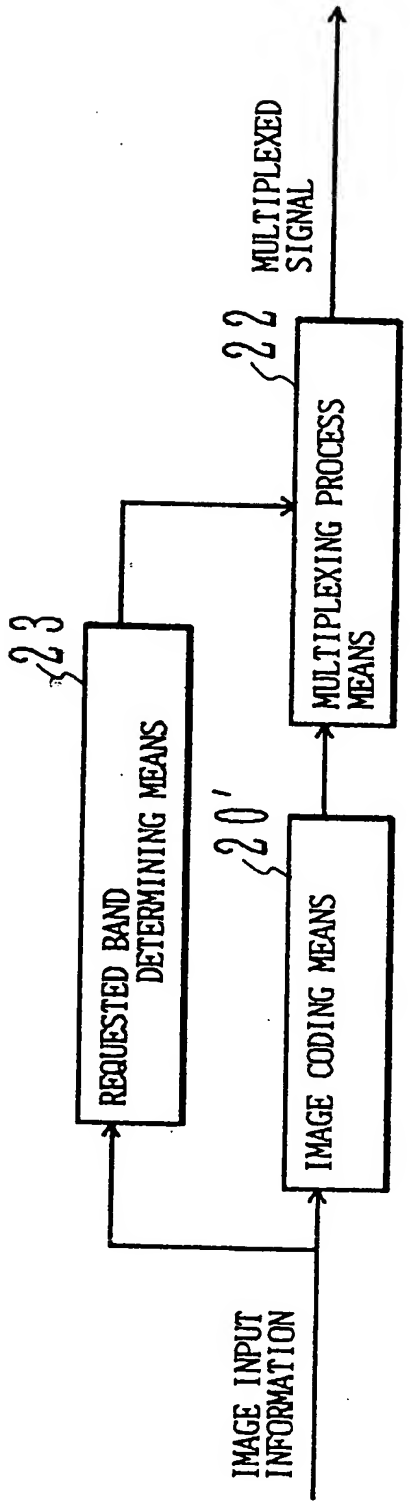
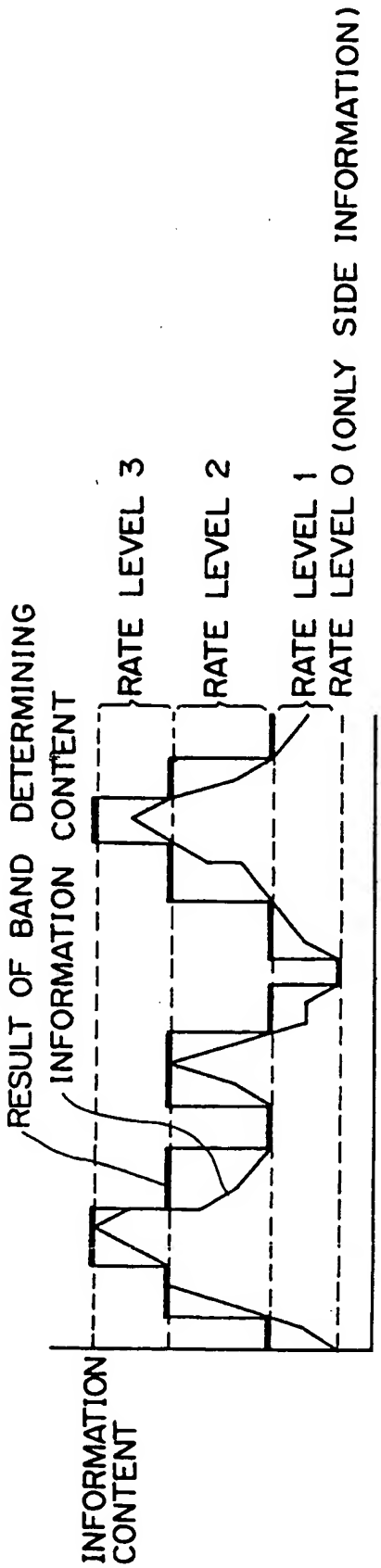


IMAGE INPUT INFORMATION
IMAGE CODING MEANS
20'
REQUESTED BAND DETERMINING MEANS
23
MULTIPLEXING PROCESS MEANS
22
MULTIPLEXED SIGNAL

Fig. 64



AN EXAMPLE FOR BAND DETERMINING

Fig. 65

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